

# MyPBX U300 Installation Guide

Version:V1.1

**Yeastar Technology Co., Ltd**

Date: 9<sup>th</sup>, Aug, 2013

# Contents

<b>1. PREPARATION BEFORE INSTALLATION .....</b>	<b>3</b>
<b>2. HARDWARE SPECIFICATIONS .....</b>	<b>4</b>
2.1 OVERVIEW .....	4
2.2 LED INDICATORS AND PORTS .....	4
2.3 SPECIFICATIONS AND OPERATING ENVIRONMENT .....	6
<b>3. MYPBX U300 INSTALLATION .....</b>	<b>7</b>
3.1 PLACEMENT INSTRUCTIONS .....	7
3.2 INSTALLATION INSTRUCTIONS .....	7
3.2.1 <i>Connection of Ethernet Ports</i> .....	7
3.2.2 <i>Connection of E1/T1 ports</i> .....	7
3.2.3 <i>Connection of FXS ports</i> .....	9
3.2.4 <i>Power Connection</i> .....	10
3.2.5 <i>Overall Flow Chart</i> .....	10
<b>4. MYPBX U300 BASIC CONFIGURATIONS .....</b>	<b>11</b>
4.1 FACTORY DEFAULTS .....	11
4.2 LOGGING IN THE WEB CONFIGURATION PANEL .....	12
4.3 NETWORK SETTINGS .....	13
4.4 EXTENSIONS SETUP AND MANAGEMENT .....	13
4.5 CONFIGURING TERMINALS .....	14
4.6 MAKING/RECEIVING CALLS .....	14
4.7 RESET TO FACTORY DEFAULTS .....	14
<b>5. CONCLUSION .....</b>	<b>15</b>

## About MyPBX U300

MyPBX U300 is a standalone embedded hybrid PBX specially designed for businesses and offices up to 100 users which is characterized by an embedded PRI(E1/T1/J1)port with up to 30 lines and 2 FXS ports in the one compact system, providing higher density trunking for offices using E1 PRI signaling. It enables the company to benefit from the latest VoIP technology and traditional reliability at the same time. Also it could be used as a gateway to legacy PBX systems in applications.

MyPBX U300 has perfect interoperability with mainstream IP Phone brands such as Yealink, Snom, Grandstream, Aastra, Polycom and Cisco. Auto provisioning of Yealink, Snom, Cisco, Aastra and Polycom IP phones would greatly save time on configurations when there is bulk extensions to register.

Besides, customers from different countries will all feel comfortable while using MyPBX U300. Because they are able to select their local language for Web GUI and Voice Prompt. Furthermore, it is renowned for its user-friendly web configuration panel which is easy accessible and self-evident making configuration a trouble-free job.

This Guide explains how to install MyPBX U300, how to configure MyPBX U300 via web interface, how to add extensions, and make/receive calls via PSTN lines.

## 1. Preparation before Installation

**Please make sure the following devices are available before installation:**

### Contents of the box

Upon receiving MyPBX U300 gift box, please open the package and check if all the items are supplied as MyPBX U300 Packing List (See Sheet 1). If there is any problem, please contact your provider.

Item	Unit	QTY	Description
MyPBX U300	PC	1	MyPBX U300 main box
Power cord	PC	1	For the input of 220V AC power
Warranty card	PC	1	With Series Number printed for Repair & Return
CD	PC	1	including all the MyPBX User Manuals and relative documents
phone line	PC	2	
Network Cable	PC	1	
Serial Line	PC	1	
Mounting Ear	PC	3	Long mounting ear: 1 PC; short mounting ear: 2 PCS
M3 Screw	PC	9	

Sheet 1 MyPBX U300 Packing List

## 2. Hardware Specifications

### 2.1 Overview



Figure 2-1 MyPBX U300

### 2.2 LED Indicators and Ports

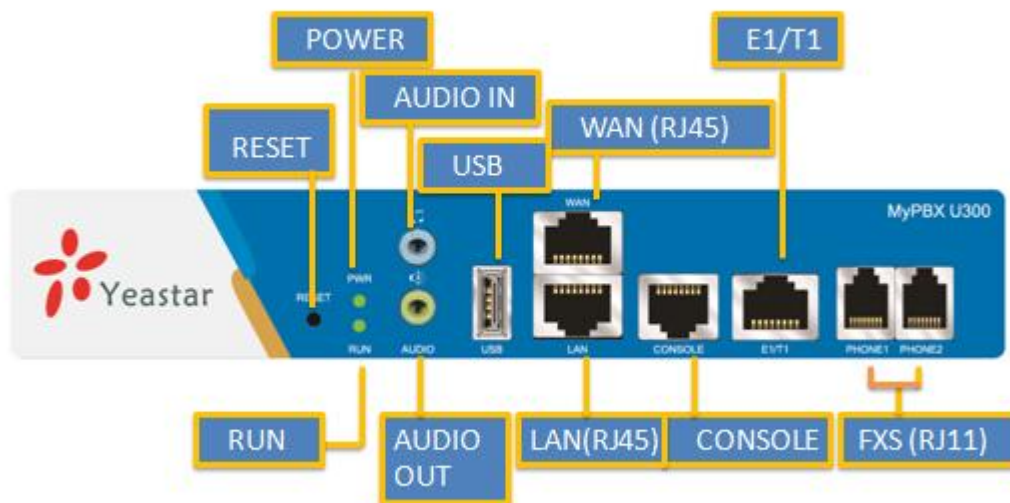



Figure 2-2 MyPBX U300 Front Panel



Figure 2-3 MyPBX U300 Rear Panel


LED	Indication	Status	Description
Power	Power status	On	The power is switched on
		Off	The power is switched off
RUN	MyPBX status	Blinking	MyPBX is running properly
		Not Blinking/Off	MyPBX goes wrong

Ports	Description
RJ11 ports	<b>FXS port:</b> For the connection of analog phones.
Ethernet ports	MyPBX U300 provides two 10/100M adaptive RJ45 Ethernet ports, marked as LAN and WAN. - <b>LAN port:</b> LAN port is for the connection to Local Area Network (LAN). - <b>WAN port:</b> WAN port is the network port for the connection to internet. It supports 'DHCP server', 'PPPoE/dynamic DNS', and "static IP" for IP address assignment.
E1/T1 ports	For the connection to E1 line or the E1 port of traditional PBX.
Audio Input port	For the connection to a music-on-hold source
Audio output port	For the connection to a paging system.
Reset Button	Press the reset button to restore the factory defaults  Please make sure that you'd like to reset because once reset, the previous configurations would be erased automatically.
Power port	For the connection of a power adapter to input 100~240V, 50~60HZ AC power
USB port	For connecting USB HDD
Console port	For debugging

## 2.3 Specifications and Operating Environment

	Description
Size	L213 * W 160 * H 44 (mm)
Power Supply	220V,50~60Hz
Operating Temperature	0°C~50°C(32°F~122°F)
Storage Temperature	-10°C~65°C(14°F~149°F)
Operating Humidity	5%~90%(Non-condensing)
Storage Humidity	0%~95%(Non-condensing)
Weight	1.2kg

## 3. MyPBX U300 Installation

 To avoid unexpected accident, personal injury or device damage, please read the following instructions before installing MyPBX U300.

### 3.1 Placement Instructions

- ◆ **Ambient Temperature:** to avoid overheating, please do not run MyPBX U300 in the place where the ambient temperature is above 122°F (50°C).
- ◆ **Ventilation:** please make sure that the device has good ventilation around.
- ◆ **Anti-jamming**—there may be some sources of interference that might affect the normal running of MyPBX U300. It's highly recommended that the device
  - Should be placed away from high-power radio, radar transmitters and high frequency, and high-current devices.
  - Is using independent power junction box and effective anti-grid interference measures have been taken.
- ◆ **Mechanical load**--Please make sure that the device is placed steadily to avoid any accident that might cause damage. If placed on the desktop, please ensure it is horizontally placed.


### 3.2 Installation Instructions

After placing MyPBX U300 in a suitable place, please connect the power adapter and all other cables to complete the installation.

#### 3.2.1 Connection of Ethernet Ports

MyPBX U300 provides two 10/100M adaptive RJ45 Ethernet ports, that is, LAN port and WAN port.

- ◆ **WAN Port Connection**  
Connect one end of a network cable to the WAN port of MyPBX U300, and the other end to the Ethernet port of a hub, switch or ADSL modem.
- ◆ **LAN Port Connection**  
Connect one end of a network cable to the LAN port of MyPBX U300, and the other end to any port of company's LAN switch/router.

 If the LAN port is connected to PC directly (not via a switch), please use cross-over cable.

#### 3.2.2 Connection of E1/T1 ports

MyPBX U300 supports one E1/T1 port. Below are the connection instructions.

##### 1. Connection of E1/T1 ports

The E1/T1 port could be connected to the E1 line or the E1 port of traditional PBX.

**Note:** the E1/T1 port of MyPBXU300 is RJ45 port, and the pins are defined as below table:

Item	Description
1	RX- Receive Data-
2	RX+ Receive Data+
3	Not used
4	TX- Transmit Data-
5	TX+ Transmit Data+
6	Not used
7	Not used
8	Not used

**Note:** Please follow the below steps to make RJ45 cable to connect it to the E1 port.

**Step 1:** Connect MyPBX U300 E1 port to the E1 line according to the line sequence of E1 port.

From the above table, we can see that the No.1 and No.2 pins of MyPBX U300 E1 port is for receiving data (No.1 for receiving data -, No.2 for receiving data +), while No.4 and No.5 is for transmitting data (No.4 is for transmitting data -, No.5 is for transmitting data +).

When connected to the E1 line, RX- of E1 port should be connected to TX- of E1 line; RX+ of E1 port should be connected to TX+ of E1 line; TX- of E1 port should be connected to RX- of E1 line; TX+ of E1 port should be connected to RX+ of E1 line.

For example, the line sequence of the E1 line (RJ45 port) provided by the telecom company is: No.4 for TX-, No.5 for TX+, No.7 for RX- and No.8 for RX+ which apparently can't correspond to that of MyPBX U300 E1 port. So you need jumper, which could be depicted as below:

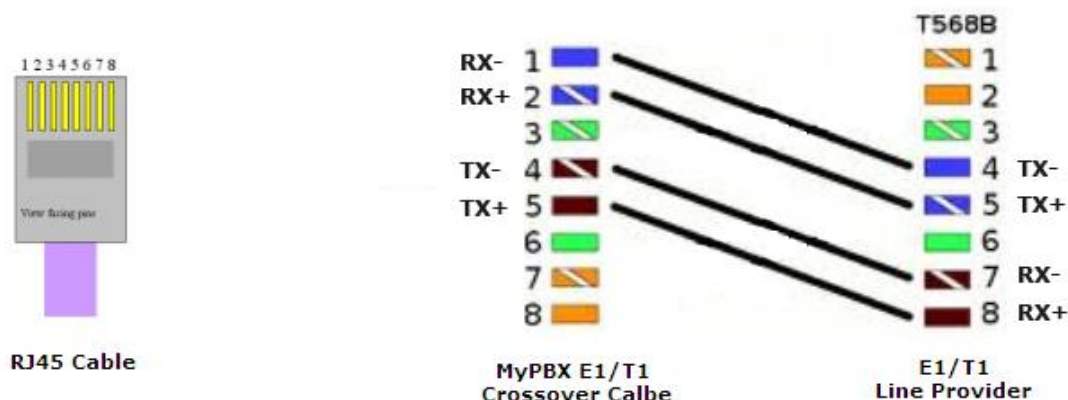


Figure 3-1 E1 cable cord sequence

**Step 2:** Connect the self-made RJ45 cable to E1 port.



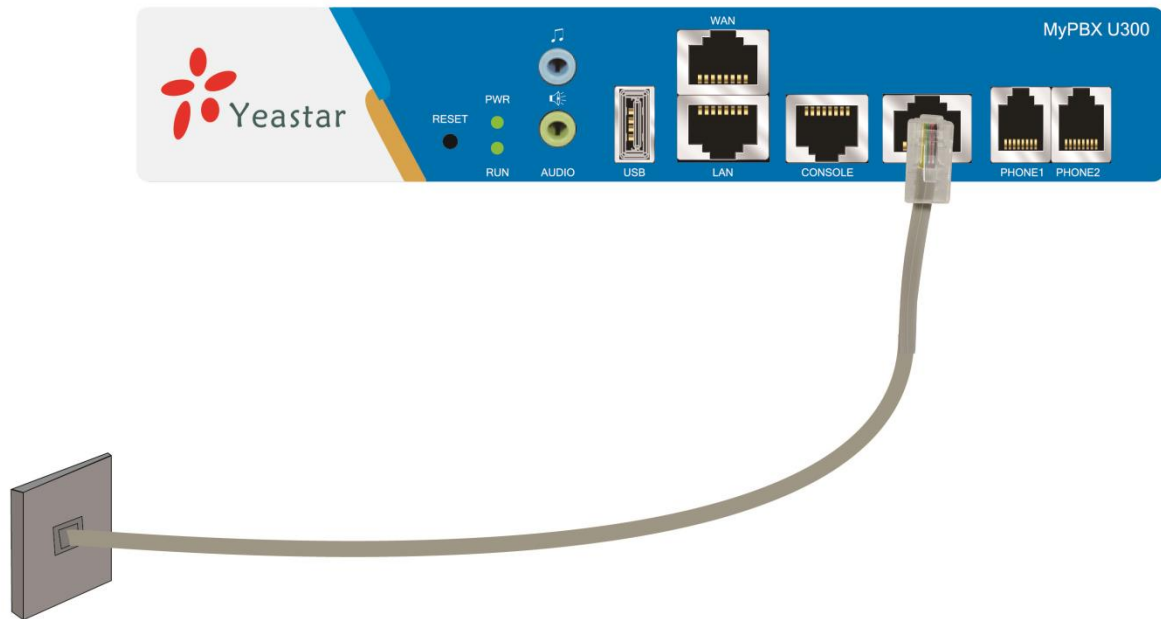


Figure 3-2 Connection of E1/T1 port

### 3.2.3 Connection of FXS ports

The FXS port could be connected to an analog phone with a phone line. The connection could be depicted as below:

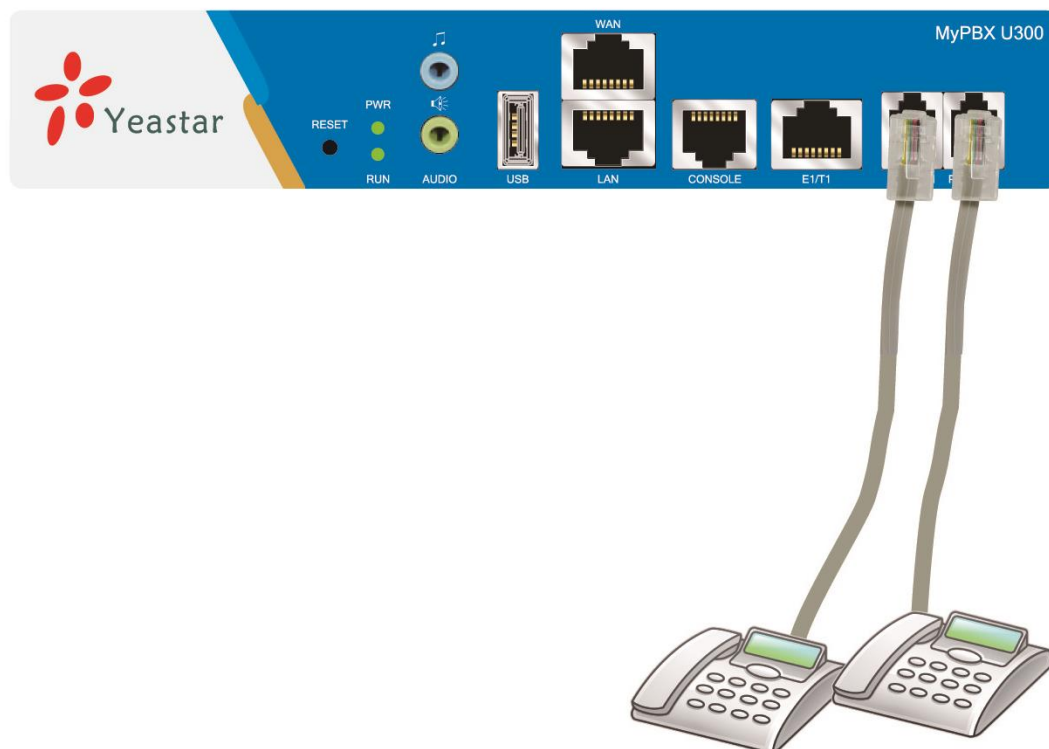


Figure 3-3 Connection of FXS ports

### 3.2.4 Power Connection

Once users have made sure that device installation, cable connection and power type is correct, please switch on the power. Then MyPBX U300 will start booting. In the meantime, users would see that the 'POWER' and 'RUN' indicator lights would turn on.

 Please switch off the power before plugging or unplugging the cables.

### 3.2.5 Overall Flow Chart

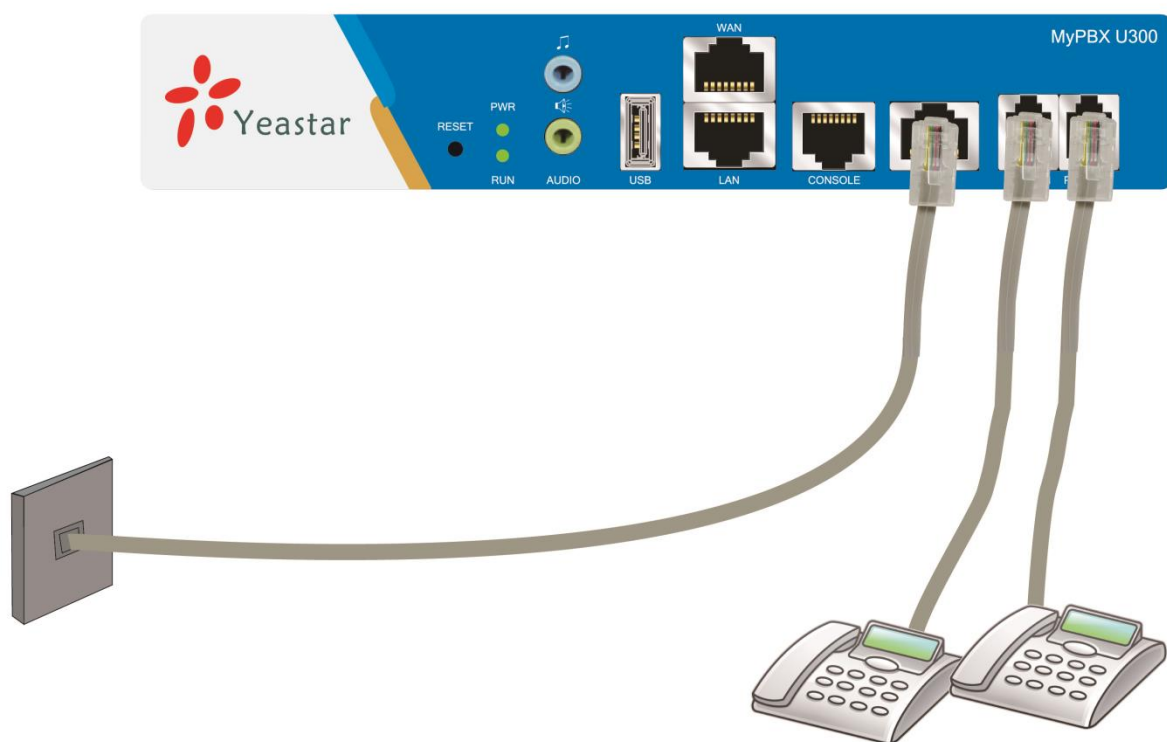


Figure 3-4 Overall Flow Chart

## **4. MyPBX U300 Basic Configurations**

### **4.1 Factory Defaults**

MyPBX U300 provides web-based configuration interface for administrator and account user. The user can manage the device by log in the web interface.

The factory default IP address:

LAN: 192.168.5.150;

Access path: [http://\[IP address\]](http://[IP address])

User Name: admin;

Password: password

Default SIP UDP port: 5060

There are 6 extensions set up by default. The user could use these extensions directly, edit or delete them. Their extension numbers are “300”-“305” (6 numbers in a row) whose passwords are “pincode+respective extension number”.

## 4.2 Logging in the Web Configuration Panel

Firstly, please check if the IP address that assigned to the network port (LAN or WAN) is in the same segment with the PC and which can be connected when commit 'Ping +MyPBX IP address' command. Start the browser on PC. In the address bar, enter the IP address, click 'enter' button and then you can see MyPBXU300 Web Configuration Panel login page (See Figure 4-1).

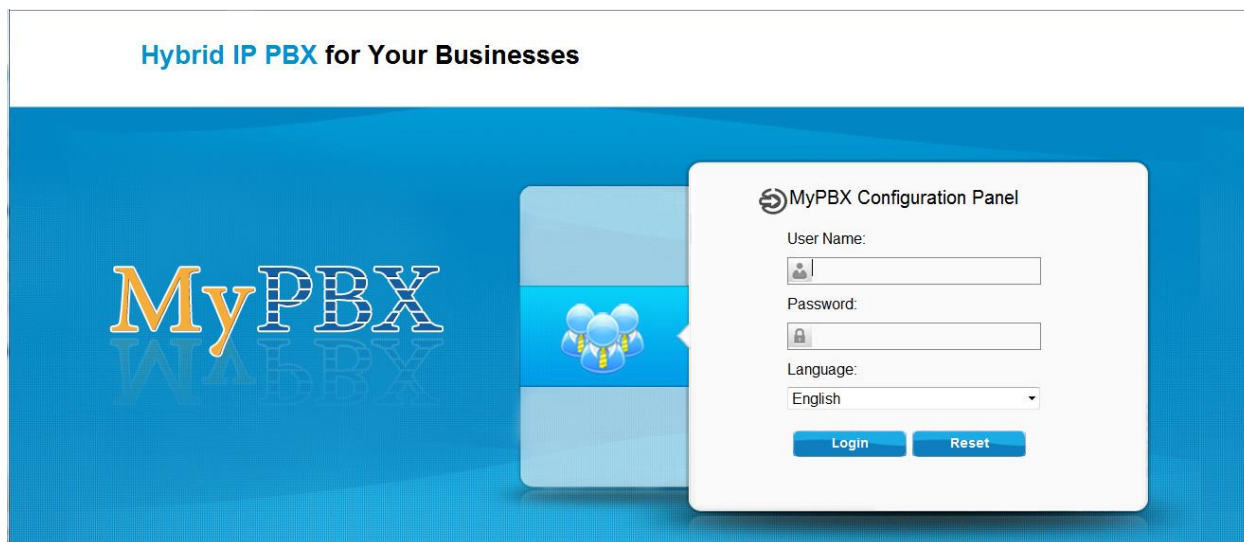


Figure 4-1 MyPBX Web Configuration Panel Login Page

Enter the Admin User Name and Password to log in.

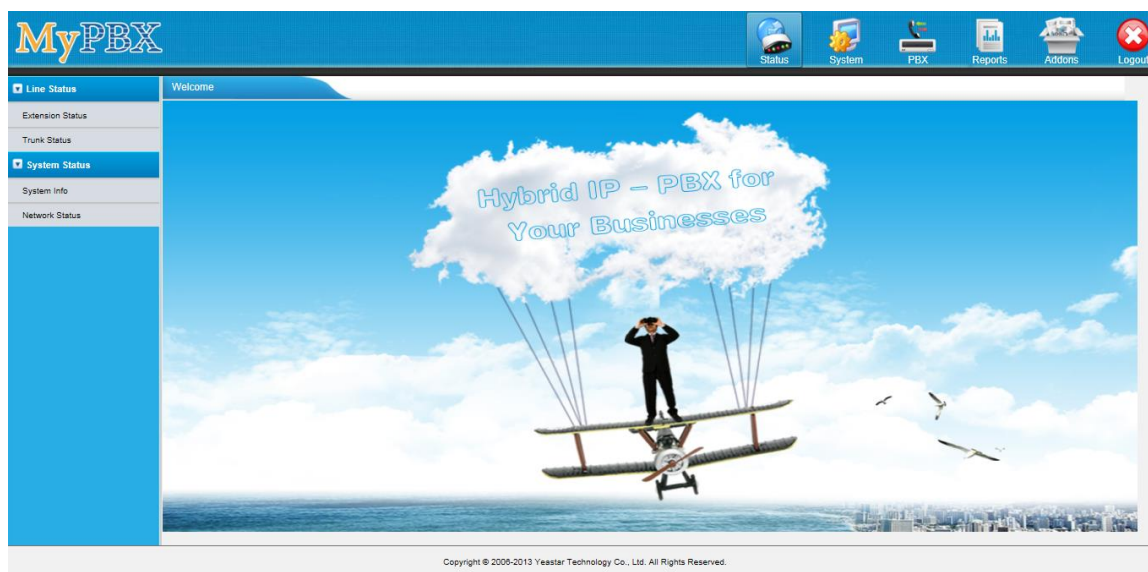


Figure 4-2 MyPBX Admin Configuration Interface

Via the configuration interface, the admin can make all the system configurations, including network settings (LAN, WAN, Firewall, VPN, DDNS, VLAN, etc.); system parameters configuration (system prompts, time zone, password, etc.); internal settings (Web access port, user extension range, business hours, conference room, paging and intercom, music on Hold, etc.); inbound routes; outbound routes; user account management (add or delete an extension account, IP restriction, features setup, etc.); extended function management (call queue, conference room, IVR, etc.); call logs (search, download); firmware update and reset, etc.

 Please note that after saving the changes, remember to click the “Apply changes” button to make the changes take effect.

## 4.3 Network Settings

After logging in the admin configuration interface, generally the first step is to configure the IP address. If both the WAN and LAN ports are connected to the network, their IP addresses need to be configured. If not, just configuring the IP address of the LAN port which is connected to the local area network of the company is OK.


### ◆ WAN Settings

Click “Network Settings”->“WAN Settings” on the left menu bar.

1. If the network that the WAN port is connected to is using DHCP server as connection method, please select the “DHCP” (dynamic IP) option.
2. If the network connection method is “Static IP address”, please select this option and enter the LAN parameters (IP address, subnet mask, default gateway, DNS, etc.)
3. If MyPBX is connected to ADSL adapter directly, please select “PPPoE” option and enter the ADSL user name and password.

### ◆ LAN Settings

Click “Network Settings”->“LAN Settings” on the left menu bar of the web configuration interface. LAN port is used for the interoperability of IP terminals and MyPBX. If the LAN port is connected to the company’s LAN, please configure the correct IP address and corresponding subnet mask.

 Please note that after changing the IP address of LAN port, MyPBX should be rebooted to make the new changes take effect.

## 4.4 Extensions Setup and management

MyPBX has already set up 6 extensions which are extensions “300-305” by default. The password is “pincode + extension number”. It’s up to the user to decide whether to directly use these default extensions or after changing the passwords. Of course, they could be deleted.

The admin could add, edit or delete the extensions via the Web Configuration Panel.

In the “Extensions” page, there are “Add Extension”, “Add Bulk Extensions”, “Delete the Selected Extension” and “Edit Extension”(configure the features such as call forwarding and voicemail) subpages, etc.


## 4.5 Configuring Terminals

MyPBX could be connected with SIP-based audio and video IP terminal devices (e.g. IP Phone, voice gateway, and soft terminal, etc.). The steps of configuring terminals are as below:

Step 1: Make sure the IP terminal and MyPBX are interconnected

Step 2: Enter the IP address of MyPBX when configuring “SIP server”, “Proxy server”, or “Registration server” of the IP terminals, and the default SIP UDP port is 5060.

Step 3: Enter the account registration information with the “extension number/password” of the default or newly setup extensions of MyPBX.

 For more information on terminal configuration, please refer to [http://www.yeastar.com/Solutions/SIP\\_Phones.asp](http://www.yeastar.com/Solutions/SIP_Phones.asp)

Step 4: Reboot the IP terminal after configuration, and click “Status Monitor”->“Line Status” on the menu bar of MyPBX web configuration interface to check if it has registered successfully to MyPBX.

## 4.6 Making/Receiving calls

Note: The MyPBX default settings are sufficient to make phone calls. It is not necessary to make any changes unless the user wants to create new extensions or trunks. Refer to Administrator’s Guide for more.

### 1. Internal calls between extensions

If two IP terminals have registered to MyPBX successfully, the user could make calls between extensions just by dialing the other’s extension number.

### 2. Outbound calls

Firstly, please connect a PSTN line to the first FXO port. Then the default extensions are able to make outbound calls via this trunk by simply dialing “9+phone number+#”. The “#” is Key as Send.

Note: The dial pattern of the default outbound route is “9.”. But if the user has changed it, please dial the new prefix instead of the digit “9” when making outbound calls.


### 3. Inbound calls

When the user calls the trunk number of the PSTN line, MyPBX would route the call to IVR (the “welcome” prompt) by default. Then the user can dial the extension number following the prompt.

## 4.7 Reset to Factory Defaults

If you forget the new IP address or the password of admin, or in other cases that you would like to restore the factory defaults, please reset the device following the below instructions:

Please press the “RESET” button located in the back panel with a paper clip or a pencil tip, then you can see all of the LED indicators in the front panel would all turn red first and turn orange seconds later. When the indicator light becomes orange, the button could be released and system begins to reset. During the process, please do not power off until the LED indicators become normal which means the reset process is completed.

 Note: After resetting, all the configurations made by the admin would be erased.

## 5. Conclusion

This Installation Guide only explains the installation and basic settings of MyPBX U300 for making and receiving calls. For more functionality and advanced settings of MyPBX U300, please refer to the relative documents as below:

***“MyPBX U300 Datasheet”***

***“MyPBX U300 Administrator’s Guide”***

***“MyPBX U300Extension User’s Guide”***

[The End]



U300

# Administrator's Guide

Version 12.18.0.22

**Yeastar Technology Co., Ltd.**



# Table of Contents

1. Introduction.....	5
1.1 Features .....	6
1.2 Hardware Specifications.....	7
1.2.1 Exterior Appearance .....	7
2. System setup.....	8
2.1 Connection Drawing .....	8
2.2 Connecting Ethernet Line .....	9
2.3 Supplying Power .....	9
3 Administrator Login .....	10
4 Status.....	11
4.1 Line status .....	11
4.1.1 Extension Status .....	11
4.1.2 Trunk Status .....	12
4.2 System Status.....	13
4.2.1 System Info.....	13
4.2.2 Network Status .....	13
5 System .....	14
5.1 Network Preferences .....	14
5.1.1 LAN Settings.....	14
5.1.2 WAN Settings.....	15
5.1.3 DHCP Server.....	16
5.1.4 VLAN Settings.....	17
5.1.5 VPN Settings.....	19
5.1.6 DDNS Settings .....	19
5.1.7 Static Route.....	20
5.2 Firewall Settings .....	22
5.2.1 Firewall Rules.....	22
5.2.2 IP blacklist.....	24
5.3 System Preference .....	26
5.3.1 Password Settings .....	26
5.3.2 Date and Time .....	27
5.3.3 Firmware Update.....	28
5.3.4 Backup and Restore .....	29
5.3.5 Reset and Reboot .....	29
5.3.6 Hot Standby .....	30
5.3.7 AMI Settings.....	31
5.3.8 Alert Settings.....	32
5.3.9 Database Grant.....	35
5.3.10 Security Center .....	36
5.4 Storage Management .....	37
5.4.1 USB Device.....	37
5.4.2 External Storage .....	37

---

6 PBX .....	40
6.1 Extensions .....	40
6.1.1 FXS/VoIP Extensions .....	40
FXS Extensions .....	40
VoIP Extensions .....	45
6.1.2 Phone Provisioning .....	50
6.2 Trunks .....	59
6.2.1 Physical Trunk .....	59
6.2.2 VoIP Trunk .....	63
6.3 Outbound Call Control .....	70
6.3.1 Outbound Routes .....	70
6.3.2 Speed Dial Settings .....	73
6.4 Inbound Call Control .....	74
6.4.1 IVR .....	74
6.4.2 Ring Groups .....	77
6.4.3 Queues .....	78
6.4.4 Conferences .....	82
6.4.5 Inbound Routes .....	82
6.5 Audio Settings .....	88
6.5.1 Custom Prompts .....	88
6.5.2 Music on Hold Prompts .....	89
6.5.3 System Prompts Settings .....	90
6.6 Basic Settings .....	91
6.6.1 General Preferences .....	91
6.6.2 Business Hours .....	93
6.6.3 Feature Codes .....	95
6.6.4 Voicemail Settings .....	98
6.7 Advanced Settings .....	101
6.7.1 SIP Settings .....	101
6.7.2 IAX Settings .....	107
6.7.3 Blacklist .....	108
6.7.4 Callback Settings .....	109
6.7.5 DNIS Settings .....	111
6.7.6 DISA .....	111
6.7.5 PIN User Settings .....	112
6.7.8 Paging Groups .....	114
6.7.9 Certificates .....	116
7 Reports .....	117
7.1 Call Logs .....	117
7.2 System Logs .....	118
8 Addons .....	119
8.1 Call Recording .....	119
8.2 MyPBX Client .....	119
8.3 Billing System .....	120

---

8.4 Hotel Module .....	120
9 Logout .....	121
10. Use MyPBX.....	122
10.1 Make outbound call .....	122
10.1.1 Sample Routing via E1 Trunk.....	122
10.1.2 Sample Routing via VoIP Trunk .....	123
10.2 Incoming call.....	126
10.2.1 Sample Routing to an IVR.....	126
APPENDIX A FAQ.....	128
APPENDIX B How to Configure Autobackup .....	129
APPENDIX C How to Configure NAT setting .....	131
APPENDIX D How to Use Auto Provision .....	133
APPENDIX E How Do I Configure Distinctive Ring Tones .....	137
APPENDIX F How to Use DID .....	139
APPENDIX G MyPBX Security configuration Guide .....	143
1. Security Configuration for Web GUI.....	143
1.1 Change the default access port for HTTP on Options page .....	143
1.2 Change the default password for the web GUI .....	144
2. Disable SSH on LAN Settings Page .....	144
2.1 Disable SSH.....	144
2.2 Change the default password for SSH .....	145
3. Security Configuration for Extensions .....	146
3.1 Change the default SIP Port.....	146
3.2 Disable guest calls.....	147
3.3 Security Configuration for remote extensions.....	148
3.4 Set a robust password and enable IP restriction for extensions .....	148
4. Setup Proper Firewall Rules.....	149
5. Alert Settings .....	156
5.1 IPATTACK .....	157
5.2 WEBLOGIN .....	158
APPENDIX H How to Enable Call Recording on MyPBXU300 .....	159
1. Activate Call Recording .....	159
2. Login .....	160
3. USB Devices .....	160
4. Recording Settings.....	161
5. Record Logs .....	162
6. Share Settings .....	163
7. Password Settings .....	165

# 1. Introduction

## **MyPBX—IP-PBX for Medium Businesses/Home Office**

MyPBX U300 is a standalone embedded hybrid PBX for medium businesses and remote branch offices of larger organizations (1-100 users per site). MyPBX U300 also offers a hybrid solution (a combination of VoIP applications using E1/T1/J1 equipment) alternative for enterprises who are not yet ready to migrate to a complete VoIP solution.

### **FCC Statement**

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radiofrequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at his own expense.

### **FCC Caution**

Any changes or modifications not expressly approved by the party responsible for compliance could void the user's authority to operate this equipment.

### **FCC Warning**

This equipment has been tested and found to comply with the regulations for a Class B digital device, pursuant to Part 15 of the FCC Rules.

Operation is subject to the following two conditions:

- (1) This device may not cause harmful interference.
- (2) This device must accept any interference received including interference that may cause undesired operation.

## 1.1 Features

• Auto-provision	• Follow me
• Audio In/Out	• Interactive Voice Response (IVR)
• BLF Support	• Intercom / Zone Intercom
• Blacklist	• Music On Hold
• Call transfer	• Music On Transfer
• Call Detail Records(CDR)	• Paging / Zone Paging
• Call Forward	• PIN Users
• Call Parking	• Queue
• Call Recording	• QOS
• Call Pickup	• Ring Group
• Call Routing	• Route by Caller ID
• Call Transfer	• Spy functions
• Call Waiting	• Skype Integration (Skype Connect)
• Caller ID	• Three-way Calling
• Call Back	• Mobility Extension
• Conference	• External Storage
• Speed Dial	• DDNS
• Define Office Time	• OpenVPN
• Direct Inward System Access(DISA)	• T.38
• DIDs	• Voicemail
• Distinctive Ringtone	• VLAN
• Do Not Disturb(DND)	• WAN
• Dial by Name	• PPPoE
• Firewalls	• Static Route
• Alert	• Multiple administrators
• Integrated built-in packet capture tools	• Blind Transfer
• Security Center	• Database Grant
• Addons, including recording/MyPBX client/Billing/Hotel	

More info, please click: <http://www.yeastar.com/Products/MyPBX-U300.asp>

## 1.2 Hardware Specifications

### 1.2.1 Exterior Appearance

#### Front Panel



Figure1-1 MyPBX U300 Front Panel

No.	Indication
①	Reset button: Press until all LEDs turn orange, which means the reset is completed.
②	Power LED: Green stably Run LED: Green Blinking
③	Audio in interface Audio out interface
④	USB 2.0 port
⑤	WAN port LAN port
⑥	Console port (RJ45)
⑦	E1/T1 interface
⑧⑨	Two fixed FXS ports(RJ11)

## 2. System setup

### 2.1 Connection Drawing

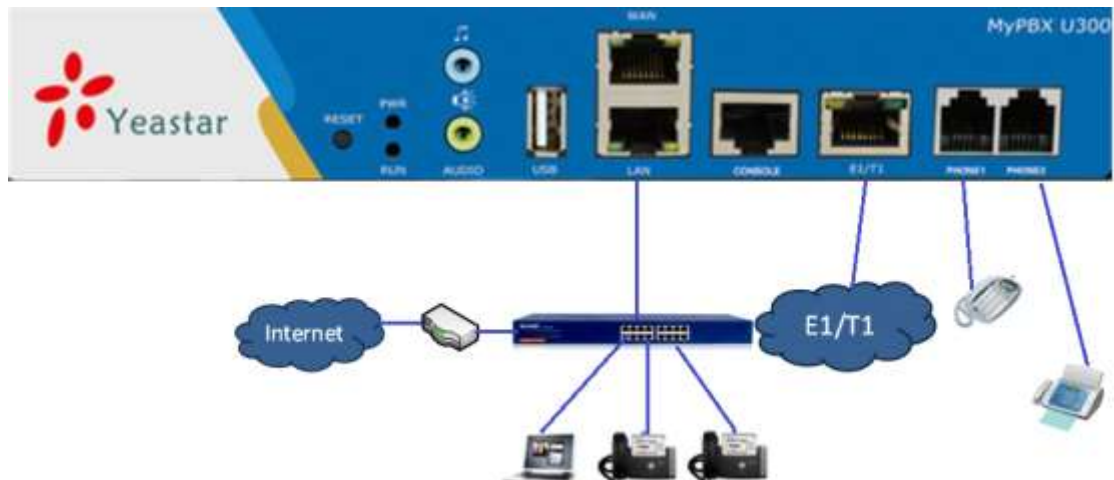


Figure 2-1 U300 Connection Drawing

## 2.2 Connecting Ethernet Line

MyPBX provides two 10/100MEthernet ports with RJ45 interface and LED indicator. Plug Ethernet line into MyPBX's Ethernet port, and then connect the other end of the Ethernet line with a hub, switch, router, LAN or WAN. Once connected, check the status of the LED indicator. A yellow LED indicates the port is in the connection process, and a green LED indicates the port is properly connected.

## 2.3 Supplying Power

Please follow the steps below to connect the MyPBX unit to a power outlet:

1. Connect the small end of the power cable to the power input port on the MyPBX back panel, and plug the other end of the cable into a 100V~240V AC power outlet.
2. Check the Power LED on the front panel. A solid green LED indicates that power is being supplied correctly.



## 3 Administrator Login

From your web browser, input the IP address of the MyPBX server.

If this is the first time you are configuring MyPBX, please use the default settings below (your PC should be in the same local network with MyPBX):

IP Address: <http://192.168.5.150>

Username: **admin**

Password: **password**

In this example, the IP address is 192.168.5.149



Figure 3-1 Login Page


This is the welcome page of MyPBX U300 after successful login.



Figure 3-2 Welcome Page

## 4 Status



Click  to start to check the status of MyPBX U300, where we can check the status of extension, trunk, network and system information.

### 4.1 Line status

In this page, we can check the status of extension and trunks






#### 4.1.1 Extension Status



Figure 4-1 Extension Status Page

#### MyPBX Status Description:

##### Extensions:

- 1)  : Extension is unavailable
- 2)  : Extension is idle
- 3)  : Extension is ringing
- 4)  : Extension is busy
- 5)  : Extension is on hold

## 4.1.2 Trunk Status



Status	Signal	Trunk Name	Type	User Name	Port/PhoneNum/IP	Inachability
Registered		Yeastar	SIP	504	192.168.4.142	OK
OK (2 ms)		SupportTel	SP-SIP		192.168.4.141	OK (2 ms)

Type	Connect	Alarm
E1		

Figure 4-2 Trunk Status Page

### Trunks:

#### VOIP Trunk:

##### Status

Unregistered: Trunk registration failed.

Registered: Successful registration, trunk is ready for use.

Request Send: Registering.

Waiting: Waiting for authentication.

#### Service Provider:







##### Status

OK: Successful registration, trunk is ready for use.

Unreachable: The trunk is unreachable.

Failed: Trunk registration failed.

#### E1 Trunk:

Connect	Alarm	Comment
		The port doesn't have E1 line connected.
		E1 line has been connected to the port; PRI is in corresponding or can't communicate.
		Succeed.

## 4.2 System Status

In this page, we can check the status of MyPBX system, including the hardware, firmware version and the network status of LAN and WAN ports .

### 4.2.1 System Info

In this page, we can check the hardware/firmware version, or the disk usage of MyPBX.



Figure 4-3 System Information

### 4.2.2 Network Status

In this page, the IP address of LAN and WAN port will appear, if OpenVPN is configured well, they will be displayed here too.



Figure 4-4 Network Status

## 5 System



Click  to access.

In this page, we can configure the network settings, firewall settings, storage management and some other preferences like firmware update and hot standby.

### 5.1 Network Preferences

#### 5.1.1 LAN Settings

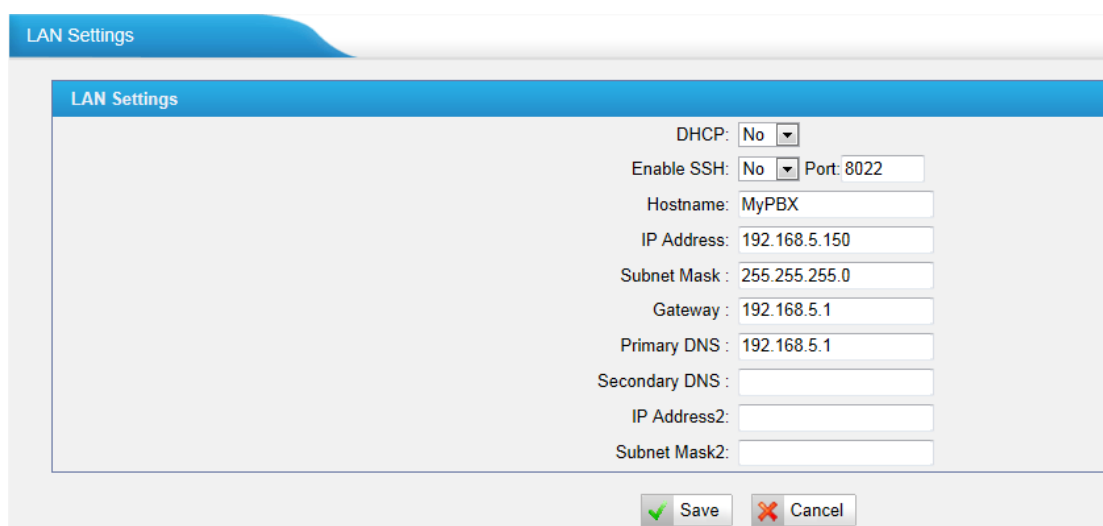


Figure 5-1 LAN Settings

#### •DHCP

If this option is set, MyPBX will use DHCP to get an available IP address from your local network. Not recommended or you cannot access MyPBX without the right IP address

#### •Enable SSH

This is the advance way to access the device, you can use the putty software to access the device. In the SSH access, you can do more advanced settings and debug. Disabled by default.

•**Port:** the default is 8022; you change it to another one

#### •Hostname

Set the host name for MyPBX.

#### •IP Address

Set the IP Address for MyPBX.

Recommend to configure a static IP address for MyPBX

#### •Subnet Mask

Set the subnet mask for MyPBX.

#### •Gateway

Set the gateway for MyPBX.

#### •Primary DNS

Set the primary DNS for MyPBX.

#### •Secondary DNS

Set the secondary DNS for MyPBX.

#### •IP Address2

Set the second IP Address for MyPBX.

#### •Subnet Mask2

Set the second subnet mask for MyPBX.

## 5.1.2 WAN Settings

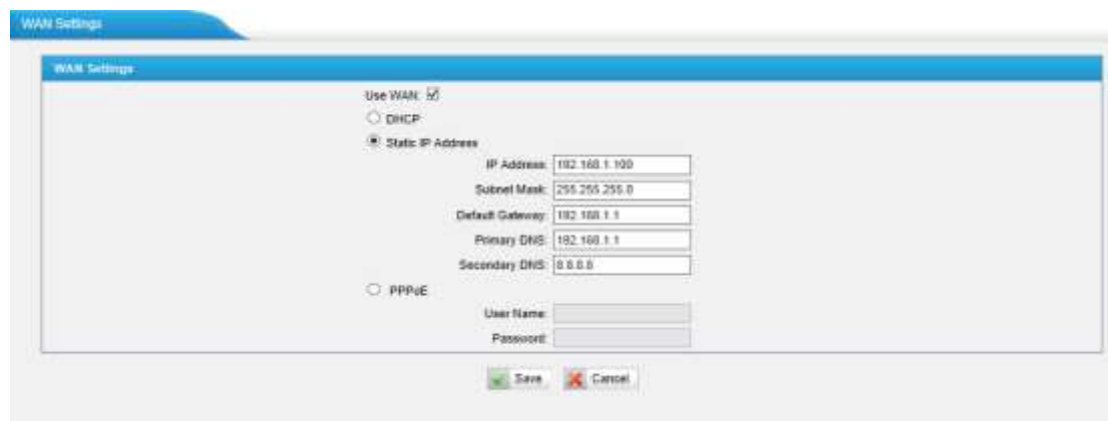


Figure 5-2 WAN Settings

It supports three connection types: DHCP (obtain an IP automatically), PPPoE, Static IP Address.

#### Notes:

1. WAN port is disabled by default.

2. WAN port cannot be used as a router to route the internet packages from WAN port to LAN port.

#### •DHCP

.If your ISP says that you are connecting through DHCP or a dynamic IP address from your ISP, perform these steps:

- Step1: Select **DHCP** as the WAN Connection Type.
- Step2: Click **Save** button to save the settings.
- Step3: Reboot the device.
- Step4: Check the WAN's Status (Status→ Network status).

#### •Static IP Address

If your ISP says that you are connecting through a static or fixed IP address from your ISP, perform these steps:

- Step1: Select **Static IP Address** as the WAN Connection Type.
- Step2: Enter the IP Address.
- Step3: Enter the Subnet Mask.
- Step4: Enter the Gateway Address.
- Step5: Enter the Primary DNS and Secondary DNS.
- Step6: Click the **Save** button to save the settings.
- Step7: Reboot the device.
- Step8: Check the WAN's Status (Status→ Network status).

#### •PPPoE

If your DSL provider says that you are connecting through PPPoE or if you normally enter a user name and password to access the Internet, perform these steps:

- Step1: Select **PPPoE** as the WAN Connection Type.
- Step2: Enter the User Name.
- Step3: Enter the Password.
- Step4: Click the **Save** button to save the settings.
- Step5: Reboot the device.
- Step6: Check the WAN's Status (Status→ Network status)

### 5.1.3 DHCP Server

Dynamic Host Configuration Protocol (DHCP) is a network protocol that enables a server to automatically assign an IP address to a computer from a defined range of numbers (i.e., a scope) configured for a given network. You can set a local network NTP server for MyPBX here too

**Note:** MyPBX U300 can work as a DHCP server, but cannot be regarded as a router. When using 'Phone Provisioning' for Grandstream IP phone, enter the IP

address of the server directly, e.g.192.168.5.150; for other phones use the default configuration.



Figure 5-3 DHCP Server Settings

## 5.1.4 VLAN Settings

A VLAN(Virtual LAN) is a logical local area network (or LAN) that extends beyond a single traditional LAN to a group of LAN segments, given specific configurations.

### Note:

MyPBX U300 is not the VLAN server, a 3-layer switch is still needed, please configure the VLAN information there first, then input the details in MyPBX, so that the packages via MyPBX will be added the VLAN label before sending to that switch.



Figure 5-4 VLAN Settings

1) VLAN Over Lan

### •NO.1



Click the NO.1 you can edit the first VLAN over Lan.

**•VLAN Number**

.The VLAN Number is a unique value you assign to each VLAN on a single device.

**•VLAN IP Address**

Set the IP Address for MyPBX VLAN over Lan.

**•VLAN Subnet Mask**

Set the Subnet Mask for MyPBX VLAN over Lan.

**•Default Gateway**

Set the Default Gateway for MyPBX VLAN over Lan

**•NO.2**

Click the NO.2 you can edit the first VLAN over Lan.

**•VLAN Number**

.The VLAN Number is a unique value you assign to each VLAN on a single device.

**•VLAN IP Address**

Set the IP Address for MyPBX VLAN over Lan.

**•VLAN Subnet Mask**

Set the Subnet Mask for MyPBX VLAN over Lan.

**•Default Gateway**

Set the Default Gateway for MyPBX VLAN over Lan.

2) VLAN Over Wan

**•NO.1**

Click the NO.1 you can edit the first VLAN over Wan.

**•VLAN Number**

.The VLAN Number is a unique value you assign to each VLAN on a single device.

**•VLAN IP Address**

Set the IP Address for MyPBX VLAN over Wan.

**•VLAN Subnet Mask**

Set the Subnet Mask for MyPBX VLAN over Wan.

**•Default Gateway**

Set the Default Gateway for MyPBX VLAN over Wan.

**•NO.2**

Click the NO.2 you can edit the first VLAN over Wan.

**•VLAN Number**

.The VLAN Number is a unique value you assign to each VLAN on a single device.

**•VLAN IP Address**

Set the IP Address for MyPBX VLAN over Wan.

**•VLAN Subnet Mask**

Set the Subnet Mask for MyPBX VLAN over Wan.

**•Default Gateway**

Set the Default Gateway for MyPBX VLAN over Wan.

## 5.1.5 VPN Settings

A virtual private network (VPN) is a method of computer networking--typically using the public internet--that allows users to privately share information between remote locations, or between a remote location and a business' home network. A VPN can provide secure information transport by authenticating users, and encrypting data to prevent unauthorized persons from reading the information transmitted. The VPN can be used to send any kind of network traffic securely. MyPBX supports OpenVPN.

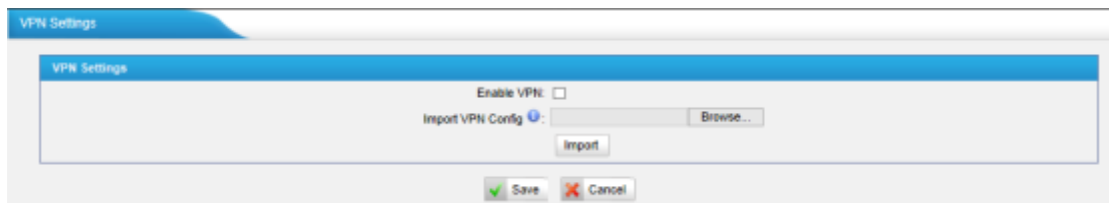


Figure 5-5 OpenVPN Settings

**•Enable VPN****•Import VPN Config**

Import configuration file of OpenVPN.

**Note:** Don't configure 'user' and 'group' in the 'config' file.  
You can get the config package from the OpenVPN provider.

## 5.1.6 DDNS Settings

DDNS(Dynamic DNS) is a method / protocol / network service that provides the capability for a networked device, such as a router or computer system using the Internet Protocol Suite, to notify a Domain Name System (DNS) name server to change, in real time, the active DNS configuration of its configured hostnames, addresses or other information.



Figure 5-6 DDNS Settings

#### •Enable DDNS

#### •DDNS Server

Select the DDNS server you sign up for service.

#### •User Name

User name the DDNS server provides you. This field is allowed to input 63 characters.

#### •Password

User account's password .

#### •Host Name

**Note:**DDNS allows you to access your network using domain names instead of IP address.The service manages changing IP address and updates your domain information dynamically.You must sign up for service through dyndns.org, freedns.afraid.org, www.no-ip.com, www.zoneedit.com

## 5.1.7 Static Route

MyPBX will have more than one internet connection in some situations but it has only one default gateway. You will need to set some Static Route for MyPBX to force it goes out through different gateway when access to different internet. The default gateway priority of MyPBX from high to low is OpenVPN→WAN port→LAN port.

Static Route Settings				
Routing Table				
Destination	Subnet Mask	Gateway	Metric	Interface
192.168.1.0	255.255.255.0	0.0.0.0	0	WAN
192.168.4.0	255.255.254.0	0.0.0.0	0	LAN
224.0.0.0	224.0.0.0	0.0.0.0	0	LAN
0.0.0.0	0.0.0.0	192.168.1.1	0	WAN

Static Route Rules				
Destination: <input type="text"/>	Subnet Mask: <input type="text"/>	Gateway: <input type="text"/>	Metric: <input type="text"/>	Interface: <input type="text"/> <input type="button" value="Add"/>
No Static Routes Defined				

Figure 5-7 Static Route Settings Page

## 1) Route table

The current route rules of MyPBX

### •Destination

The destination network to be accessed to by MyPBX

### •Subnet Mask

Specify the destination network portion.

### •Gateway

Define which gateway MyPBX will go through when access to the destination network.

### •Metric

The cost of a route is calculated by using what are called routing metric. Routing metrics are assigned to routes by routing protocols to provide measurable statistic which can be used to judge how useful (how low cost) a route is.

### •Interface

Define which internet port to go through.

## 2) Static Route Rules

You can add new static route rules here.

## 5.2 Firewall Settings

Firewalls are used to prevent unauthorized Internet users from accessing private networks connected to the Internet, especially intranets. All messages entering or leaving the intranet pass through the firewall, which examines each message and blocks those that do not meet the specified security criteria.

### 5.2.1 Firewall Rules

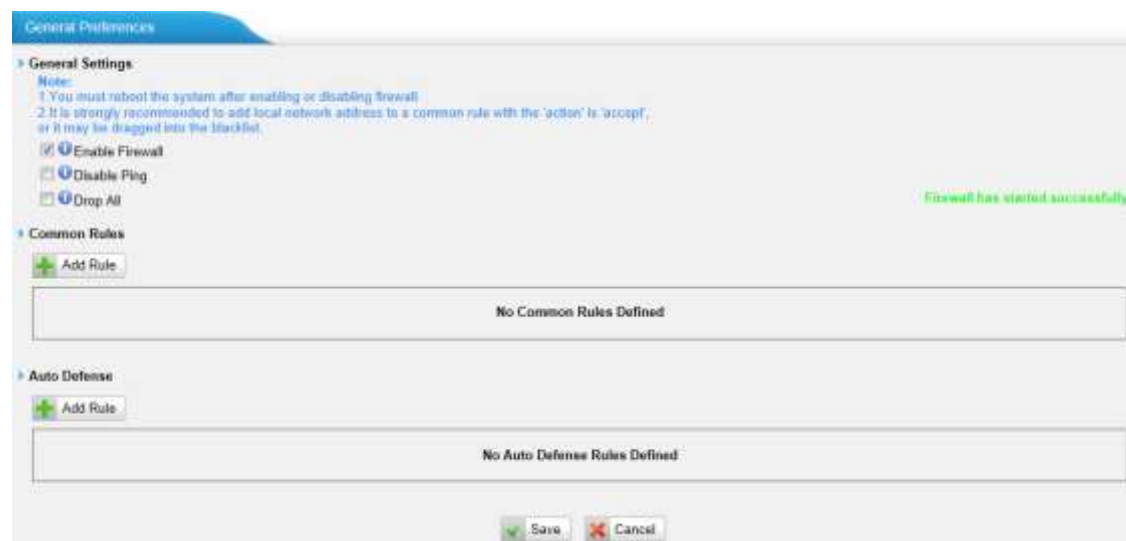


Figure 5-8 Firewall Settings

#### 1) General Settings

##### •Enable Firewall

Enable the firewall to protect the device. You should reboot the device to make the firewall run successfully.

##### •Disable Ping

Enable this item, net ping from remote hosts will be dropped.

##### •Drop All

When you enable 'Drop All' feature, system will drop all packets or connection from other hosts if there are no other rules defined. To avoid locking the devices, at least one 'TCP' accept common rule must be created for port used for SSH access, port used for HTTP access and port sued for CGI access.

#### 2) Common Rules

There is not default rules, you can create them as required

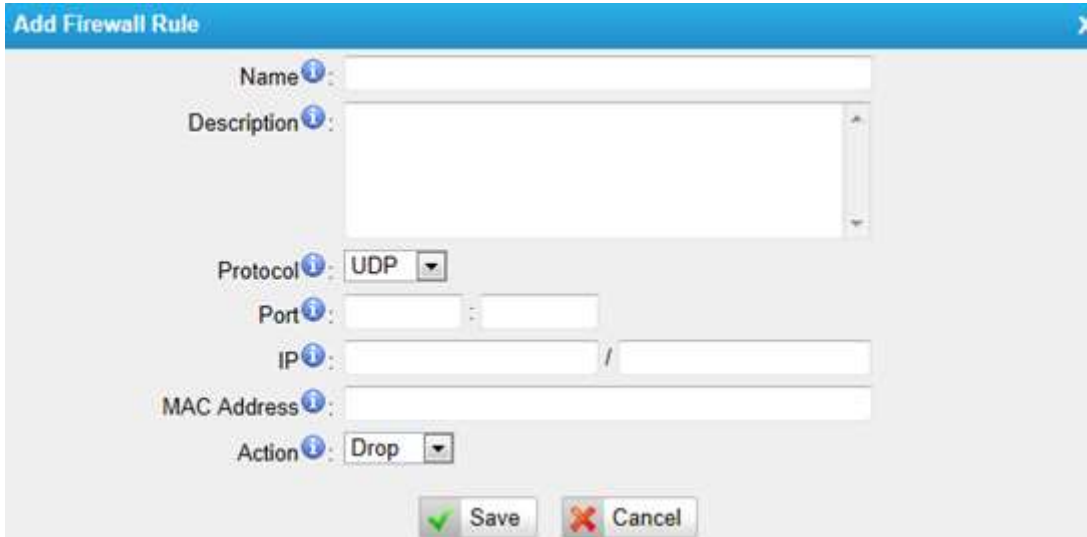


Figure 5-9 Add Common Rules

#### •Name

A name for this rule, e.g. 'HTTP'.

#### •Description

Simple description for this rule. E.g.: Accept the specific host to access the web interface for configuration.

#### •Protocol

The protocols for this rule.

#### •Port

Initial port should be on the left and end port should be on the right.  
The end port must be equal to or greater than start port.

#### •IP

The IP address for this rule. The format of IP address is: IP/mask

Ex:192.168.5.100/255.255.255.255forIP 192.168.5.100

Ex:216.207.245.47/255.255.255.255 for IP 216.207.245.47

Ex:192.168.5.0/255.255.255.0 for IP from 192.168.5.0 to 192.168.5.255.

#### •MAC Address

The format of MAC Address is XX:XX:XX:XX:XX:XX, X means 0~9 or A~F in hex, the A~F are not case sensitive.

**Note:** The MAC address will be changed when it's remotely device, so it will not be working to filter using MAC for remote devices.

#### •Action

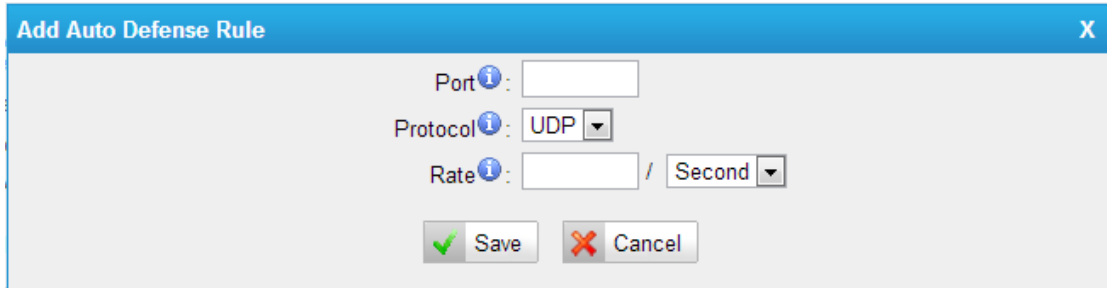
Accept: Accept the access from remote hosts.

Drop: Drop the access from remote hosts.

Ignore: Ignore the access.

### 3) Auto Defence

By default, there is no rule.



The dialog box titled "Add Auto Defense Rule" contains the following fields and controls:

- Port:** A text input field.
- Protocol:** A dropdown menu currently set to "UDP".
- Rate:** A text input field followed by a unit dropdown menu currently set to "Second".
- Buttons:** "Save" (with a green checkmark icon) and "Cancel" (with a red X icon).

Figure 5-10 Add Auto Defense Rule

**Port**

The port you want to auto defense, for example, 8022.

**Protocol:**

Select the protocol. You can select UDP or TCP.

**Rate:**

The maximum packets or connections can be handled per unit time.

For example, if you configure it as below:

Port: 8022

Protocol: TCP

Rate: 10/min

Then, it means maximum 10 TCP connections can be handled in 1 minute. The 11<sup>th</sup> connection will be dropped.

## 5.2.2 IP blacklist

You can set some packets accept speed rules here. When a IP address which hasn't been accepted in common rules sends packets faster than the allowed speed, it will be set as black IP address and blocked automatically.



The IP Blacklist Settings Page includes the following sections:

- Blacklist Rules:** A table with columns for Port, Protocol, and Rate. It contains three rules:
 

Port	Protocol	Rate
5000	UDP	120/s
5002	UDP	40/s
8022	TCP	500/s
- IP Blacklist:** A section showing "No Auto Black IP Address".

Figure 5-11 IP Blacklist Settings Page

## 1) Blacklist rules

We can add the rules for IP blacklist rate as your demand

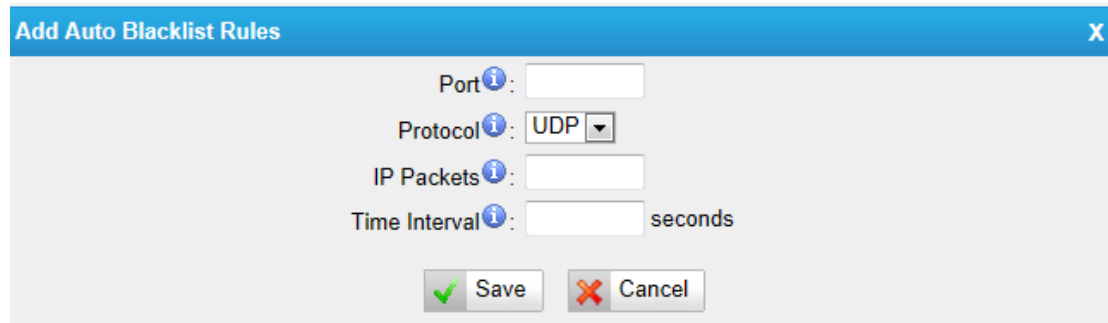


Figure 5-12 Add Blacklist Rule

### •Port

Auto defense port

### •Protocol

Auto defense protocol. TCP or UDP.

### •IP Packets

Allowed IP packets number in the specific time interval.

### •Time interval

The time interval to receive IP packets. For example, IP packets 90,time interval 60 means 90 IP packets are allowed in 60 seconds.

## 2) IP blacklist

The blocked IP address will display here, you can edit or delete it as your wish.



## 5.3 System Preference

In this page, we can set other system preference, like the password for admin account, system date and time, firmware update, hot standby, backup and restore, reset and reboot.

### 5.3.1 Password Settings

MyPBX has 3 accounts: admin, user, and cdr. User and cdr account is disabled by default.

Admin account:

The default password for account "admin" is "password". To change the password, select "admin" in "User", enter the old password and new password, click "Save". The system will then prompt you to re-login using your new password.

After you enter the new password, MyPBX will prompt the password strength. It is recommended that you use numbers, upper-case letters, and lower-case letters to increase the security.

When you log in MyPBX using "admin" account, you can enable "user" and "cdr" account; also, you can change their passwords.



Figure 5-13 Modify admin's Password

User account:

User account is disabled by default and its default password is "password". When enabling "user" account for the first time, MyPBX will ask you to change "user" password. If you don't change it, you can't enable "user" account. To change the password, select "user" in "User", enter the old password and new password, click "Save". The system will then prompt you to re-login using your new password.

After you enter the new password, MyPBX will prompt the password strength. It is recommended that you use numbers, upper-case letters, and lower-case letters to increase the security.



The screenshot shows the 'Change Password' and 'User Setting' sections of the MyPBX administrator interface. In the 'Change Password' section, the 'User' dropdown is set to 'user'. Below it are three password fields: 'Enter Old Password', 'Enter New Password', and 'Retype New Password', all containing masked characters. A green 'Strong' indicator is visible next to the new password field. In the 'User Setting' section, the 'Enable User Account' dropdown is set to 'Yes' and the 'Enable CDR Account' dropdown is set to 'No'. A 'Save' button is at the bottom.

Figure 5-14 Modify user's Password

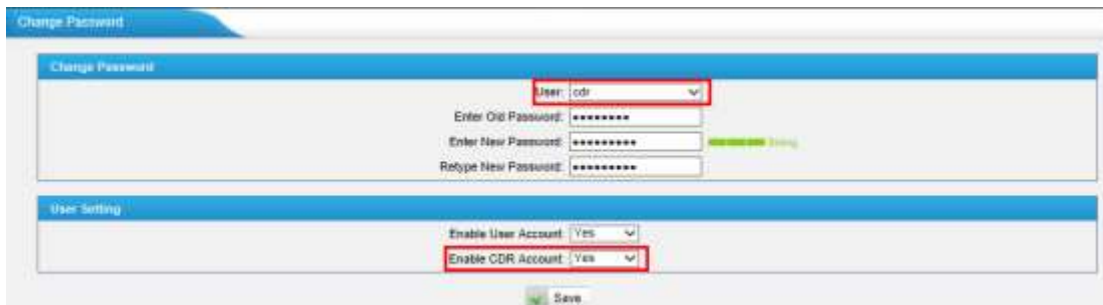
After enabling "user" account, you can log in MyPBX using "user". "user" account can change its own password.

CDR account:

"cdr" account is disabled by default and its default password is "password". You can enable it after you log in MyPBX using "admin" account.

To change the password, select "cdr" in "User", enter the old password and new password, click "Save". The system will then prompt you to re-login using your new password.

After you enter the new password, MyPBX will prompt the password strength. It is recommended that you use numbers, upper-case letters, and lower-case letters to increase the security.



The screenshot shows the 'Change Password' and 'User Setting' sections of the MyPBX administrator interface. In the 'Change Password' section, the 'User' dropdown is set to 'cdr'. Below it are three password fields: 'Enter Old Password', 'Enter New Password', and 'Retype New Password', all containing masked characters. A green 'Strong' indicator is visible next to the new password field. In the 'User Setting' section, the 'Enable User Account' dropdown is set to 'Yes' and the 'Enable CDR Account' dropdown is set to 'Yes'. A 'Save' button is at the bottom.

Figure 5-15 Modify cdr's Password

After enabling "cdr" account, you can log in MyPBX using "cdr". "cdr" account can change its own password.

## 5.3.2 Date and Time

Set the date and time for MyPBX.

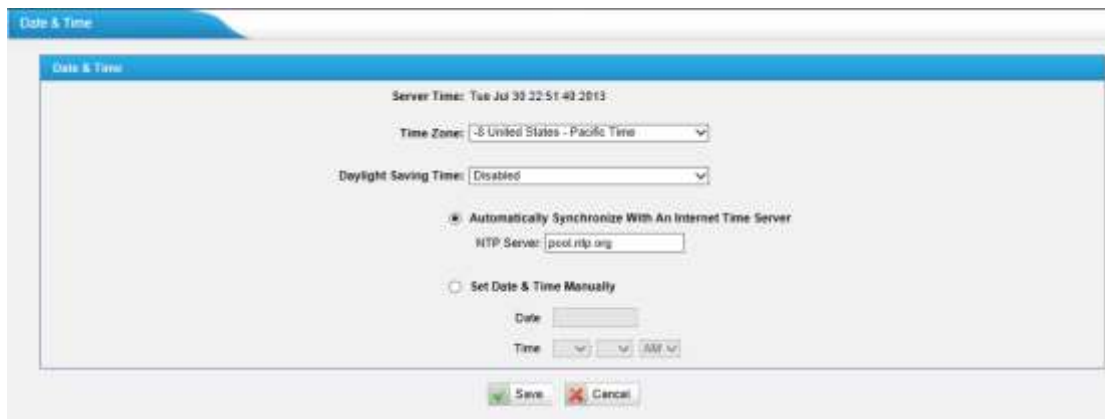


Figure 5-16 Configuring Date&amp;Time

#### •Time Zone

You can choose your time zone here.

#### •Daylight Saving Time

Set the mode to Automatic or disabled

#### •Automatically Synchronize With an Internet Time Server

Input the NTP server so that MyPBX will update the time automatically

#### •Set Date & Time Manually

You can set the time to your local right time manually here

## 5.3.3 Firmware Update

Upgrading of the firmware is possible through the Administrator web interface using a TFTP Server or an HTTP URL.

Enter your TFTP Server IP address and firmware file location, then click start to update the firmware

#### Notes:

1. If enabled "Reset configuration to Factory Defaults", System will restore to factory default settings.
2. When update the firmware, please don't turn off the power. Or the system will get damaged.
3. More information for the steps to update the firmware, please refer to this link: [http://www.yeastar.com/download/MyPBX/MyPBX\\_Standard&Pro\\_FirmwareUpFirmw\\_en.pdf](http://www.yeastar.com/download/MyPBX/MyPBX_Standard&Pro_FirmwareUpFirmw_en.pdf)

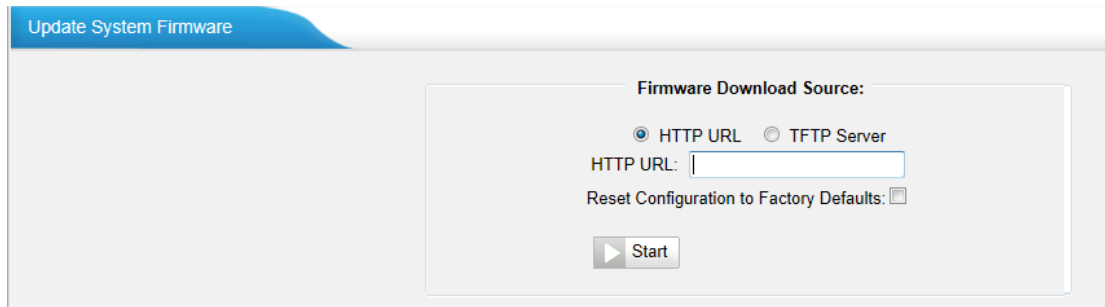


Figure 5-17 Firmware Update Page

### 5.3.4 Backup and Restore

We can backup up the configurations before reset MyPBXU300 to factory defaults, and then restore it using this package.

#### Notes:

1. Only configurations, custom prompts will be backed up, the voicemail and recording files are not included.
2. When you have updated the firmware version, it's not recommended to restore using old package.



Figure 5-18 Backup and Restore Page

### 5.3.5 Reset and Reboot

We can reset or reboot MyPBXU300 via web directly in this page.

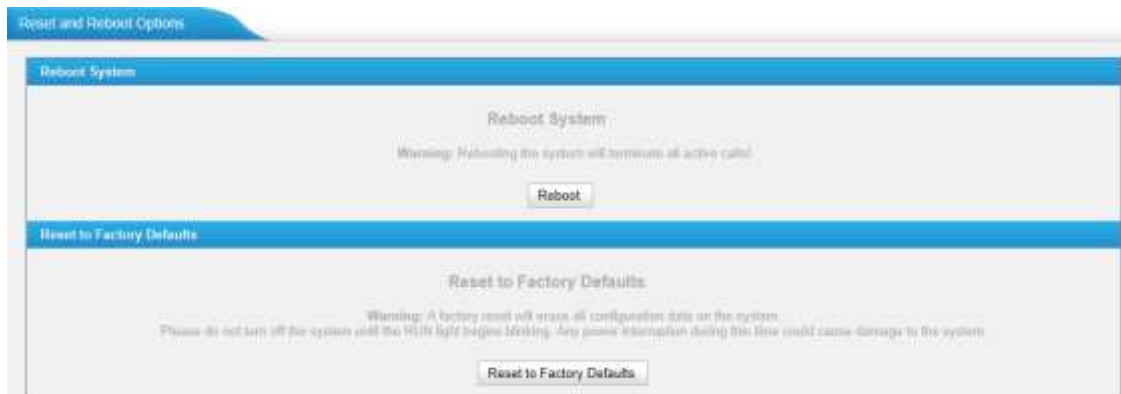


Figure 5-19 Reset and Reboot Options

## •Reboot System

**Warning:** Rebooting the system will terminate all active calls!

## •Reset to Factory Defaults

**Warning:** A factory reset will erase all configuration data on the system. Please do not turn off the system until the RUN light begins blinking. Any power interruption during this time could cause damage to the system.

## 5.3.6 Hot Standby

Redundancy is achieved by using duplicate hardware and software installations and synchronizing data and operating state. Redundancy assures smooth operation even if a system goes down. Essentially a duplicate backup system takes over with virtually no loss of service. This technique assures absolute reliability no matter what failure occurs. In mission critical installations, redundancy is a way to address possibility of any failure.

**Note 1:** Before enabling the Host Standby feature, please make sure that the two servers in the failover pair are the same model, own the same modules installed in the same slots, the same hardware configurations and firmware version.

**Note 2:** Please configure the primary server first and configure the secondary server only after the running status of primary server becomes "active".

**Note 3:** The virtual IP address inputted in this page will be the one used for registering in each IP phone.

**Note 4:** Before configuring the Email list in this page, please configure the "voicemail settings" in "PBX→Basic settings", and make sure the SMTP test successfully.

**Note 5:** Before configure the SMS list; please make sure the SIM and GSM/UMTS modules are installed well



**Hot Standby**

Note 1: Before enabling the Host Standby feature, please make sure that the two servers in the failover pair are the same model, have the same modules installed in the same slots, the same hardware configurations and firmware version.  
 Note 2: Please configure the primary server first and configure the secondary server only after the running status of primary server becomes "active".

Step 1: Verify the basic information of the server  
 Step 2: Configure the IP address and hostname of the primary and secondary servers  
 Step 3: Configure Hot Standby Settings (Example)

**Basic**

Running Status: ☒ Disabled

Enable: ☒ No

Mode:

Secondary Hostname:

Secondary IP:

Access Code:

Virtual IP Address:

Network Connection Detection:

**Basic Notification**

Notification Methods:

Email List:

**Advanced**

Heartbeat Options

Keep Alive:

Deal Time:

Figure 5-20 Hot Standby Configuration

**Mode:** Primary means the main unit; Secondary means the standby unit;

**Secondary/Primary Hostname:** If this unit mode is primary, then you need to input the hostname of standby unit; vice versa, if this unit is selected as secondary, then the hostname of primary unit is required. In brief, you need to input each other's host name on this field.

**IP:** You need to input each other's IP address on this field.

**Access code:** To make an identification number to verify each other. The number must be the same to both units.

**Virtual IP address:** To fill in a virtual IP address includes mask, which is always points to the currently activated unit. Customer can register IP phones through this virtual IP address. Please make sure the virtual IP add includes mask is the same on both units but different from their former IP address.

**Network Connection Detection:** Generally it requires the IP address of the router or gateway that connects both units. MyPBX will connect another unit through this IP address.

**Down Notification:** The way of informing customer that the system down.

**Keep Alive:** Every 2 seconds, a package will be sent from one unit to another, which can test whether they are working properly.

**Dead Time:** The default setting is 120 seconds. If there's no response within 120s after one receiving a package from the other, then the normal working unit will figure the other unit is dead and send an email or SMS to report the failure.

**Disk Synchronization:** It works for synchronizing the data on hard disk only, such as the call recording files and CDR files saved in disk. The configurations in MyPBX will not be influenced by this feature. Two options are available: timing synchronization and real-time synchronization.

### 5.3.7 AMI Settings

The Asterisk Manager Interface (AMI) is a system monitoring and management interface provided by Asterisk. It allows live monitoring of events that occur in the system, as well enabling you to request that Asterisk perform some action. The actions that are available are wide-ranging and include things such as returning status information and originating new calls. Many interesting applications have been developed on top of Asterisk that take advantage of the AMI as their primary interface to Asterisk.

There are two main types of messages on the Asterisk Manager Interface: manager events and manager actions.

The 3<sup>rd</sup> party software can work with MyPBX using AMI interface. It is disabled by default. If necessary, you can enable it.

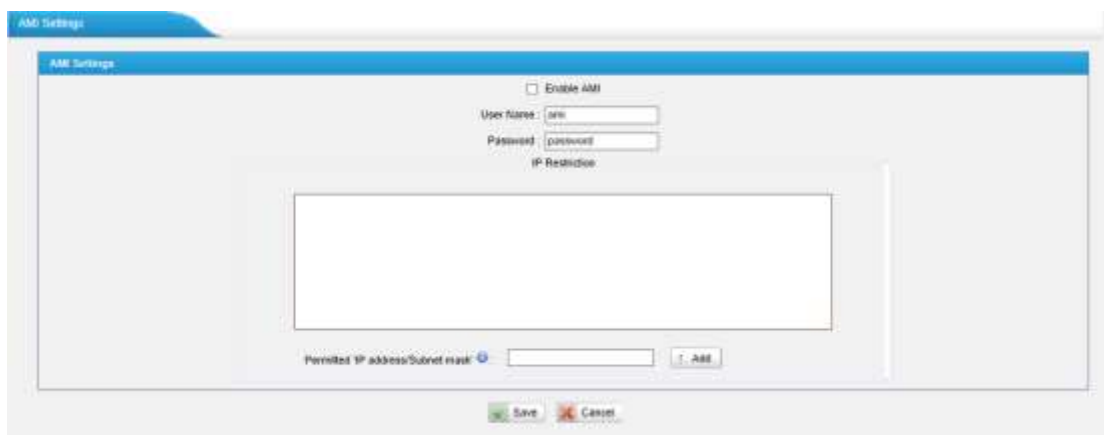


Figure 5-21 AMI Settings

**Username & password:** after enabling AMI, you can use this username and password to log in MyPBX AMI

**IP Restriction:** you can set which IP can log in MyPBX AMI interface

## 5.3.8 Alert Settings

If the device is attacked, the system will notify users the alert via call or E-mail. The attack modes include IP attack and Web Login.

More details for the system security configuration, please refer to [APPENDIX G MyPBX Security Configuration Guide](#).



Figure 5-22 Alert Settings

### 1.IPATTACK

When the system is attacked by IP address, the firewall will add the IP to auto IP Blacklist and notify the user if it match the protection rule.

#### 1) Phone Notification Settings

##### •PHONE Notification

Whether enable phone notification.

##### •Number

The numbers could be set for alert notification, users can setup multiple extension and outbound phone numbers. Please separate them by “;”.

Example: “500;9911”, if the extension has configured Follow Me Settings, the call would go to the forwarded number directly.

**•Attempts**

The attempts to dial a phone number when there is no answer.

**•Interval**

The interval between each attempt to dial the phone number. Must be greater than 3 seconds, the default value is 10 seconds.

**•Prompt**

Users will hear the prompt while receiving the phone notification.

## 2) E-mail Notification Settings

**Note:** Please ensure that all voicemail settings are properly configured on the System Settings -> Voicemail Settings page before using this feature.

**•E-mail Notification**

Whether enable E-mail Notification

**•Recipient's Name**

The recipients for the alert notification, and multiple email addresses are allowed, please separate them by “;”.

Example: jerry@yeastar.com;jason@yeastar.com, 456@sina.com .

**•Subject**

The subject of the alert email.

**•Email Content**

Text content support for predefined variables. Variable names and corresponding instructions are as follows:

\$(HOSTNAME)	Host name
\$(LOCALIP)	Local IP address
\$(SOURCEIP)	Attack source IP address
\$(DATETIME)	Occurred
\$(USERNAME)	User name (WEBLOGIN effective)
\$(DESTMAC)	Attacks destination MAC (IPATTACK effective)
\$(DESTPORT)	Attacks destination Port number (IPATTACK effective)
\$(PROTOCOL)	Protocol type (IPATTACK effective)
\$(INTERFACE)	Network interface name (IPATTACK effective)



IPATTACK
X

**Phone Notification Settings**  
Phone Notification: Yes ☐  
Number:   
Attempts: 1 ☐  
Interval:  s  
Prompt: default ☐ [Custom Prompts](#)

**E-mail Notification Settings**  
E-mail Notification: Yes ☐  
To:   
Subject:   

pbx hostname:\${HOSTNAME}

attack source ip address:\${SOURCEIP}

attack dest mac:\${DESTMAC}

attack source port:\${DESTPORT}

attack source protocol:\${PROTOCOL}

attack occurred:\${DATETIME}

Figure 5-23 IP Attack Configuration

## 2.WEBLOGIN

Web Login Alert Notification: Enter the password incorrectly five times to login MyPBX Web interface will be as attack, the system will limit the IP login within 10 minutes and notify the user.

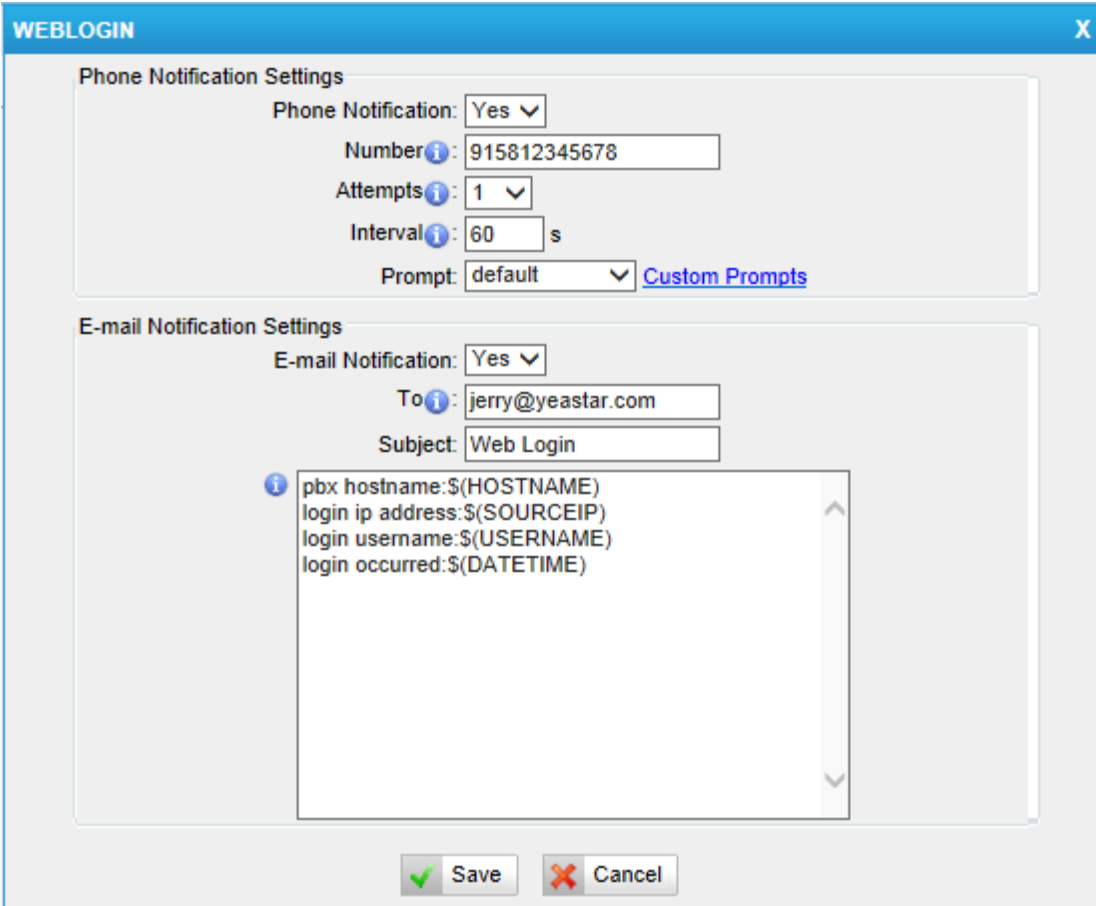


Figure 5-24 Web Login Alert Setting

### 5.3.9 Database Grant

U300 are using MySQL database from 12.18.0.22. The 3<sup>rd</sup> party software can access MySQL via internet. Before that, you need to grant the authority to the database user.

After entering "Database Grant" page, clicking "Add", you can add a database user, set user password and grant authority.

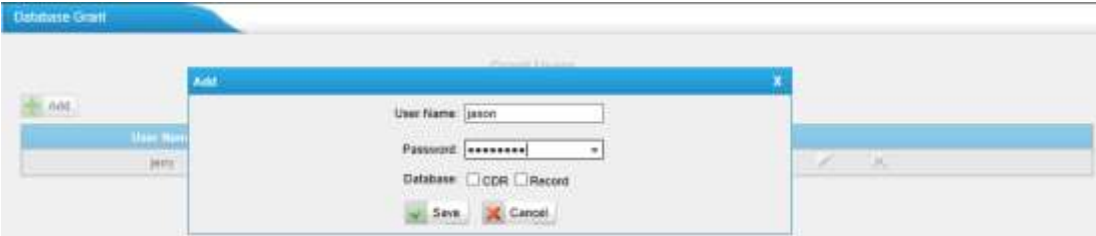


Figure 5-25 Database Grant

**Username/password:** The 3<sup>rd</sup> party can use this username and password to access the MySQL password.

**Database:** there are 2 options, CDR and Record. If you choose CDR, then this user has authority to check CDR database; if you choose Record, then the user has authority to check which call has been recorded automatically.

### 5.3.10 Security Center

You can check MyPBX security configuration in "Security Center" page. And also, you can enter the relevant security settings page rapidly.

Firewall:

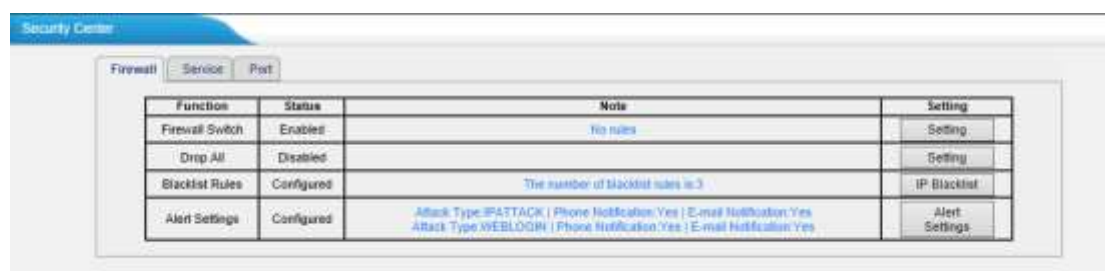


Figure 5-26 Security Center-Firewall

In the "Firewall" tab, you can check firewall configuration and alert settings. By clicking the relevant button, you can enter the configuration page directly.

Service:

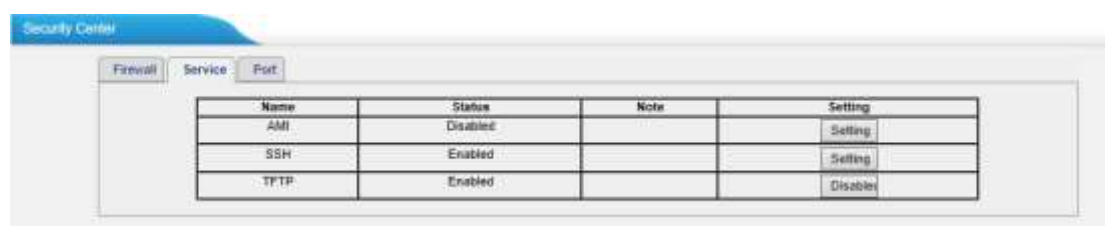


Figure 5-27 Security Center-Service

In "Service" tab, you can check AMI/SSH/TFTP status. For AMI/SSH, you can enter the according page by clicking the button in "Setting" column. For TFTP, you can directly disable or enable it.

Port:

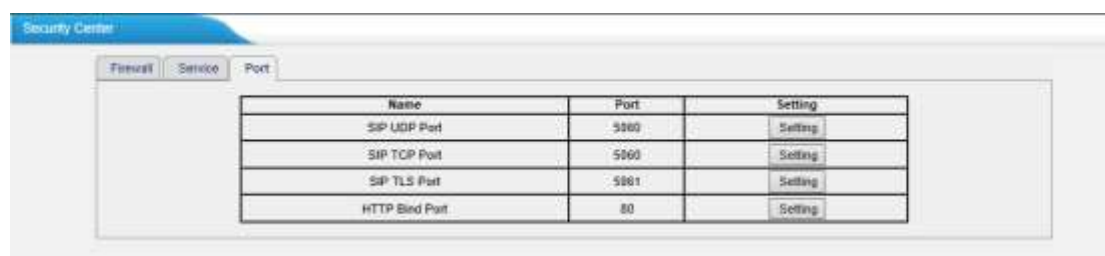


Figure 5-28 Security Center-Port

In "Port" tab, you can check SIP port and HTTP port. You can also enter the relevant page by clicking the button in "Setting" column.

## 5.4 Storage Management

### 5.4.1 USB Device

MyPBXU300 supports USB hard disk. If you enable recording add-on, you can store the recording file in USB hard disk.

You can check the information of the hard disk, manage the disk and do some related configuration.



Figure 5-29 USB Device Management

#### •USB Device Information

It's the information of the USB hard disk.

#### •USB Devices Management

You can format, clean up or remove the USB device here.

Note: Please check if you need to back up these files before formatting or clearing up the USB device. Don't power off the device during the period.

### 5.4.2 External Storage

The External Storage feature is used to extend storage space. Once configured, the files (voicemail, call recording files) created before the configured days will be moved to the Net-Disk.

**Note:** The shared folder must be based on Windows operation system.

Before external storage can be properly configured, an SMB share folder accessible from MyPBX must be set up on a Windows based machine. Once that has been set up, please follow the steps below.

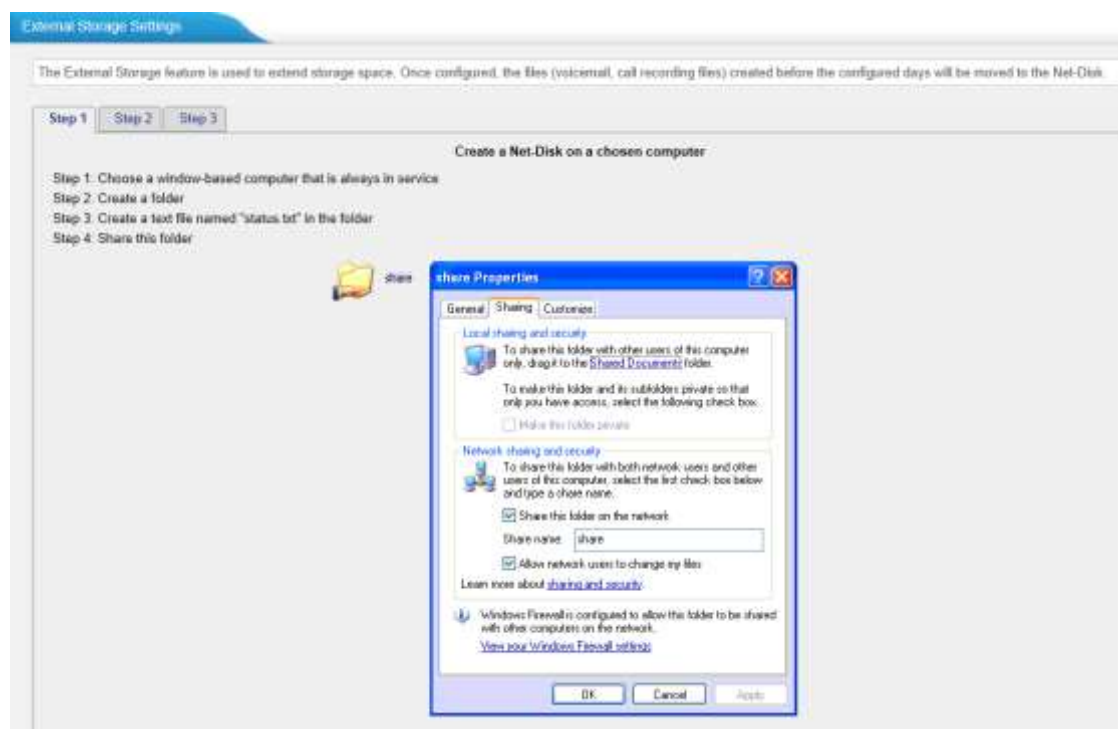


Figure 5-30 External Storage Settings

**Step 1:** Choose a window-based computer that is always in service

**Step 2:** Create a folder

**Step 3:** Create a text file named "status.txt" in the folder

**Step 4:** Share this folder

Then we need to input the Net-Disk information in Step 2 page.



Figure 5-31 Enter the Net-disk information

### Net-Disk Host/IP:

Change this to the IP address of the computer where backup files will be stored.

### Net-Disk Share Name:

Change this to the name of the shared folder where backups will be stored.

**Net-Disk Share Username:**

The user name used to log into the network share. Leave this blank if it is not required

**Net-Disk Share Password:**

The password used to log into the network share. Leave this blank if it is not required

If configuring is correctly, open your Windows share folder to see if the MyPBX backup files and folders has been created. If the contents of the backup folder look similar to step3 page, then you have successfully configured external storage on the MyPBX unit.

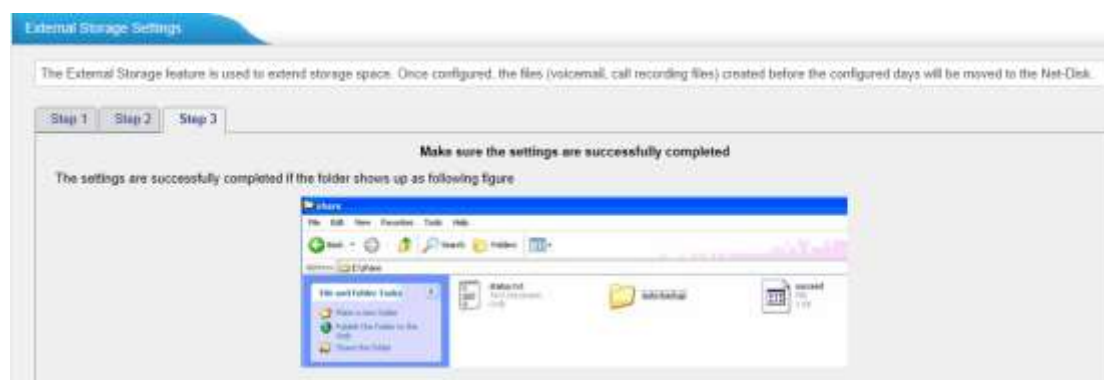


Figure 5-32 Configuring External Storage Successfully

## 6 PBX



Click  to access

In this page, we can configure the settings of extension, trunk, inbound call control, outbound call control, audio settings and the others. When configured well, we can make calls as scheduled

### 6.1 Extensions

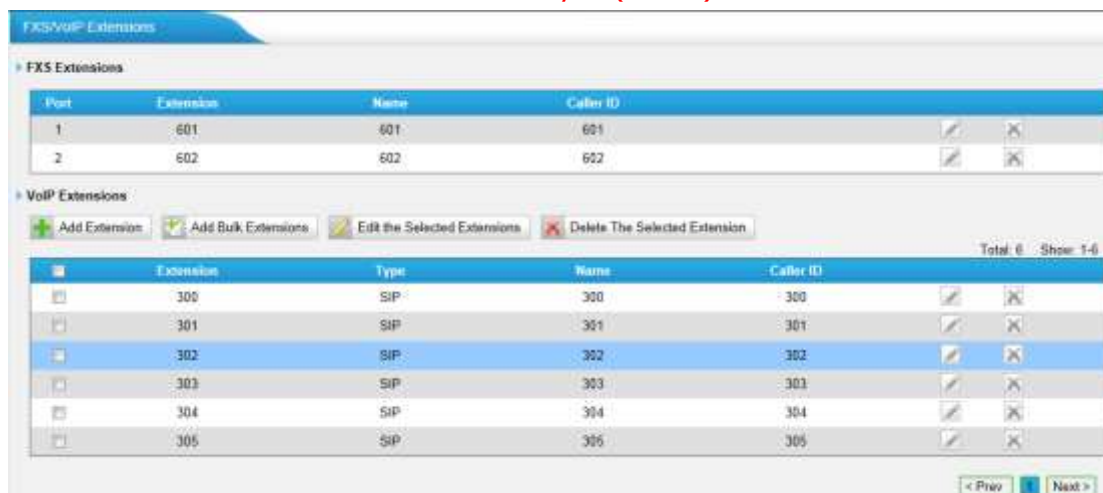
In this page, we can configure the extensions' details and provision the supported models automatically.

#### 6.1.1 FXS/VoIP Extensions

There are three types of extensions supported in MyPBX U300: SIP, IAX and analog extension.

**Note:**

1. The number of SIP/IAX extension is 100
2. The number of FXS extension is only 2 ( fixed)



The screenshot shows the 'FXS/VoIP Extensions' management interface. It is divided into two main sections: 'FXS Extensions' and 'VoIP Extensions'.

**FXS Extensions Table:**

Port	Extension	Name	Caller ID		
1	601	601	601		
2	602	602	602		

**VoIP Extensions Section:**

Buttons: Add Extension, Add Bulk Extensions, Edit the Selected Extensions, Delete The Selected Extension

Total: 6 Show: 1-6

**VoIP Extensions Table:**

	Extension	Type	Name	Caller ID		
	300	SIP	300	300		
	301	SIP	301	301		
	302	SIP	302	302		
	303	SIP	303	303		
	304	SIP	304	304		
	305	SIP	305	305		

Navigation: < Prev 1 Next >

Figure 6-1 Extension List

### FXS Extensions





FXS Extensions				
Port	Extension	Name	Caller ID	
1	601	601	601	 
2	602	602	602	 

Figure 6-2 FXS Extension List

There are two fixed analog extensions in MyPBX U300, to modify the extension number, please delete it first, then recreate it again.

### 1) General

Edit Extension - 601

General

Other Settings

General

Extension: 601

Port: 1

Name: 601

Caller ID: 601

Voicemail

☒ Enable Voicemail

Voicemail Access PIN #: 601

Mail Setting

☐ Enable Send Voicemail

Email Address:

Note: Please ensure that the section 'SMTP Settings for Voicemail'(in the 'Voicemail Settings') have been properly configured before using this feature.

Flash

Hook Flash Detection: 1000 ms

Group

Pickup Group: ---

Call Duration Setting

Max Call Duration: s

Save

Cancel

Figure 6-3 Edit FXS Extension

#### •Extension

The numbered extension, i.e. 1234, that will be associated with this particular User / Phone.

#### •Port

The extension correspond port.

#### •Name

A character-based name for this user, i.e. "Bob Jones".

#### •Caller ID



The Caller ID (CID) string will be used when this user calls another internal user.

## 2) Voicemail

### •Enable Voicemail

Check this box if the user should have a voicemail account.

### •Voicemail Access PIN #

Voicemail Password for this extension, i.e. '1234'.

## 3) Mail Setting

### •Enable Send Voicemail

Once enabled, the voicemail will be sent to the email address below as an attachment.

### •Send Voicemail to Email Address

This option defines whether or not voicemails/Fax is sent to the Email address as an attachment.

**Note:** Please ensure that all voicemail settings are properly configured on the System Settings -> Voicemail Settings page before using this feature.

## 4) Flash

### •Hook Flash Detection

Sets the amount of time, in milliseconds, that must pass since the last hook-flash event received by MyPBX before it will recognize a second event. If a second event occurs in less time than defined by Hook Flash Detection, then MyPBX will ignore the event. The default value of Flash is 1000ms, and it can be configured in 1ms increments.

## 5) Group

### •Pickup Group

If this extension belongs to a pickup group, any calls that ring this extension can be picked up by other extensions in the same pickup group by dialing the Call Pickup feature code (default \*4).

**Note:** \*4 is the default setting, it can be changed under Feature Codes -> General -> Call Pickup.

## 6) Call Duration Settings

Setup the max call duration for every call of this extension, but it's only valid for outbound calls. Enter '0' or leave this blank empty, the value would be equal to the max call duration configured in the Option Settings page.

Note: This setting will not be valid for internal calls.

## 7) Other options

### •Call Waiting

Check this option if the extension should have Call Waiting capability. If this option is checked, the 'When busy' follow me options will not be available.

### •DND

Don't Disturb.

### •User Web Interface

Check this option to allow the user to login to the MyPBX User Web interface, which can be used to access voicemail and extension recordings. Users may login to the MyPBX User Web interface by using their extension number and voicemail PIN # as the login and password respectively.

### •Ring Out

Check this option if you want to custom the ring time. Tone will stop over the time defined

## 8) Follow me (Call Forwarding)

This function sets inbound call forwarding on an extension. An administrator can configure FollowMe for this extension

## 9) Volume Settings

Rxgain: The Volume sent to FXS extension.

Txgain: The Volume sent out by the FXS extension

## 10) Mobility Extension

MyPBX allows you to use your mobile phone as extension. If you set your mobile phone as mobility extension and then you call MyPBX with this mobility phone, you will hear a dial tone. MyPBX will recognize your call as a call from an extension. You can dial the number of other extensions (Your caller ID will be the number of your extension) or dial out via outbound routes just like dialing from your extension.

**Note:** If callback is enabled in the inbound route, the mobility extension function of this inbound route will be disabled

## 11) Spy Settings

MyPBX allows extension to monitor/barge in other conversation. Once this feature is enabled, the extension has the ability to monitor/barge in other calls using the feature codes for each spy mode, refer to 'Feature Codes' page for more information.

## •spy modes

There are 4 spy modes available for choice:

General spy: you have the permission to use the following 3 modes.

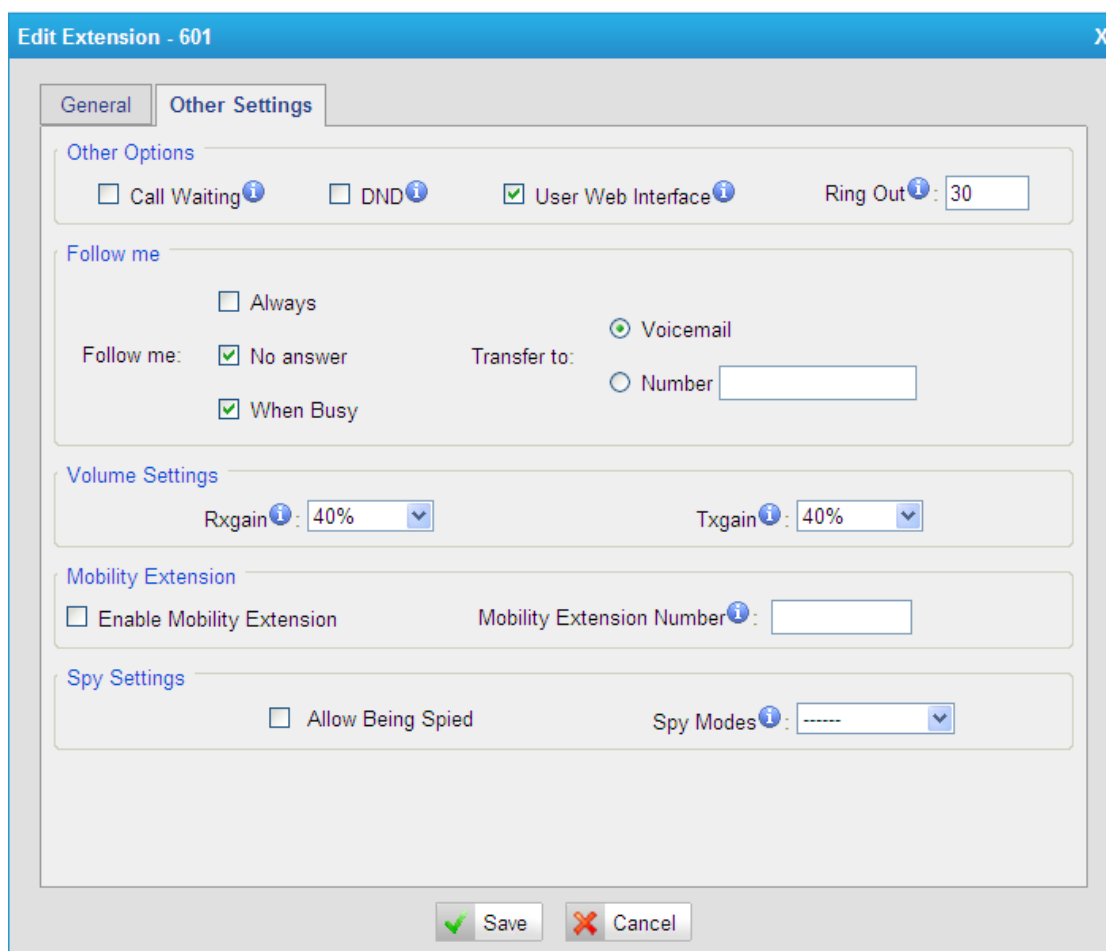
Normal spy: you can only hear the call, but can't talk

Whisper spy: you can hear the call, and can talk with the monitored extension

Barge spy: you can hear the call and talk with them both

**Note:** for example,if 500 want to monitor extension 501, we need to enable the 'allow being spied ' for 501, and choose the spy mode for extension 500. Then pick up 500 and dial "feature codes + 501" to start monitoring when 501 is in a call.

If 500 chooses 'normal spy', it should dial '\*90501' to start monitoring;  
if 500 chooses 'whisper spy', it should dial '\*91501' to start monitoring;  
if 500 chooses 'barge spy', it should dial '\*92501' to start monitor;  
if 500 chooses 'general spy', it can dial '\*90501','\*91501' or '\*92501' to start monitor.

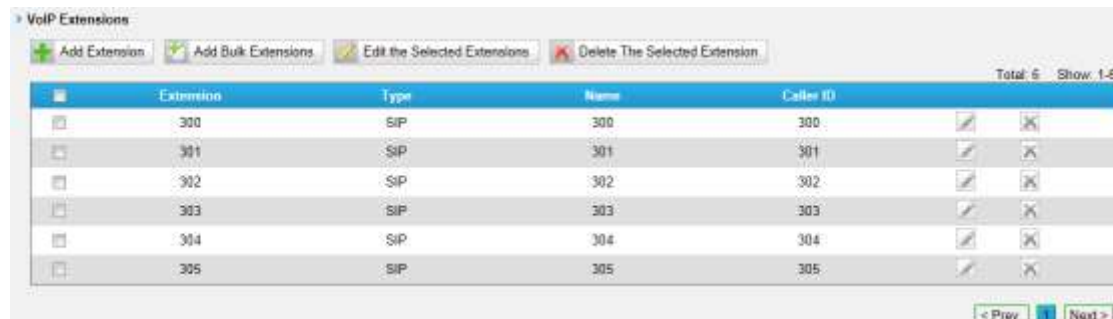


The screenshot shows the 'Edit Extension - 601' window with the 'Other Settings' tab selected. The 'Other Options' section includes checkboxes for 'Call Waiting', 'DND', and 'User Web Interface' (checked), and a 'Ring Out' field set to 30. The 'Follow me' section has a radio button for 'Always', a 'Follow me' section with 'No answer' and 'When Busy' checked, and a 'Transfer to' section with 'Voicemail' selected. The 'Volume Settings' section has 'Rxgain' and 'Txgain' both set to 40%. The 'Mobility Extension' section has an 'Enable Mobility Extension' checkbox and a 'Mobility Extension Number' field. The 'Spy Settings' section has an 'Allow Being Spied' checkbox and a 'Spy Modes' dropdown menu. At the bottom are 'Save' and 'Cancel' buttons.

Figure 6-4 FXS Extension Other Settings

## VoIP Extensions

A VOIP extension is a SIP/IAX Account that allows an IP Phone or an IP Soft-Phone client to register on MyPBX



Extension	Type	Name	Caller ID
300	SIP	300	300
301	SIP	301	301
302	SIP	302	302
303	SIP	303	303
304	SIP	304	304
305	SIP	305	305

Figure 6-5 VOIP Extension List

We can click 'Add extension' to start.

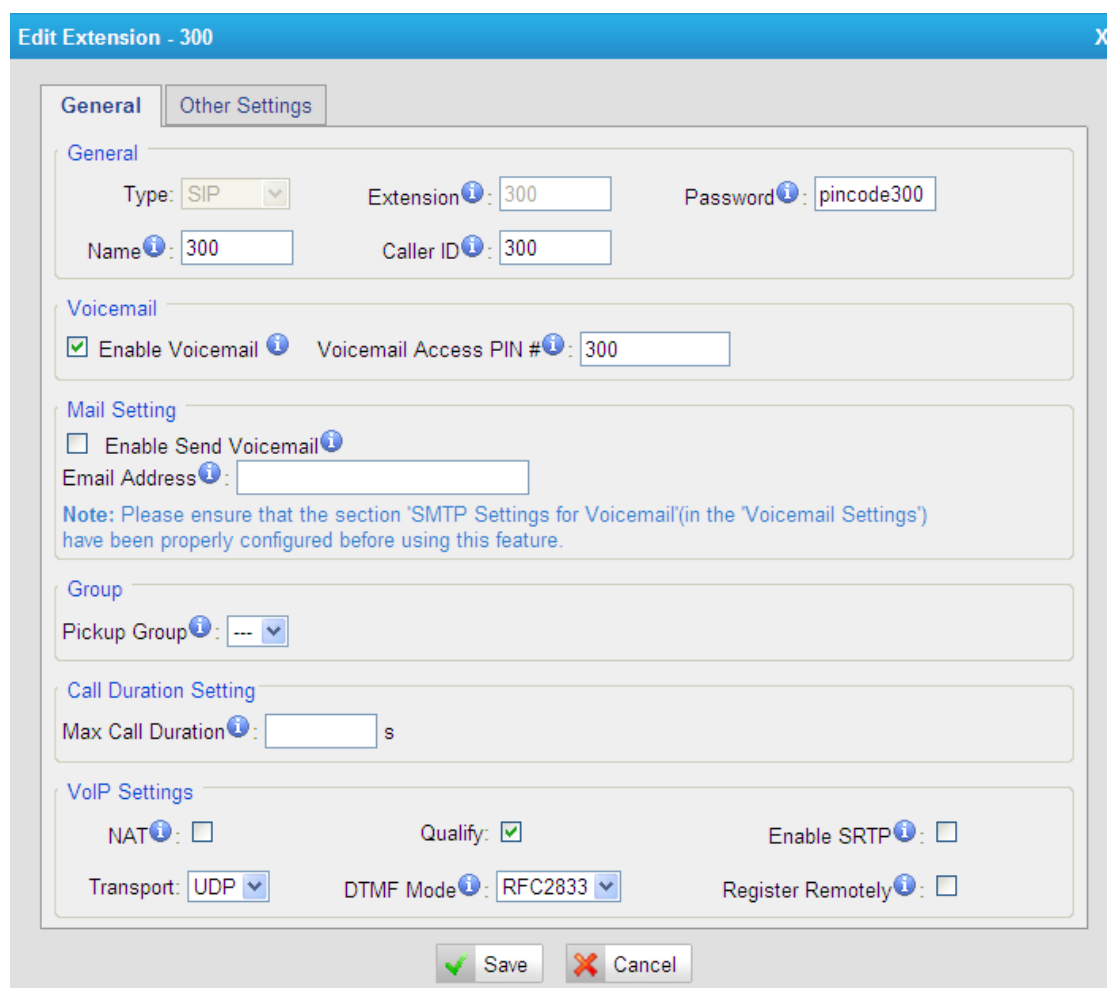


Figure 6-6 Add/Edit VOIP Extension

1) General

### •Type

Extension type: SIP,IAX or SIP/IAX.

SIP – The extension sends and receives calls using the VoIP protocol SIP.

IAX -The extension sends and receives calls using the VoIP protocol IAX.

**•Extension**

The numbered extension, i.e. 1234, that will be associated with this particular User / Phone.

**•Password**

The password for this extension, e.g. "12t3f6"

**•Name**

A character-based name for this user, e.g. "Bob Jones"

**•Caller ID**

The Caller ID will be used when this user calls another internal extension.

2) Voicemail

**•Enable Voicemail**

Check this box if the user should have a voicemail account.

**•Voicemail Access PIN #**

The voicemail Password for this extension, i.e. '1234'.

3) Mail Setting

This option defines whether or not voicemails or faxes are sent to an Email Address as attachment.

**•Enable Send Voicemail**

Once enabled, the voicemail will be sent to email as an attachment.

**•Email Address**

Email address used to receive the voicemail or Fax.

**Note:** Please ensure that the section 'SMTP Settings For Voicemail'(in the 'Voicemail Settings') have been properly configured before using this feature.

4) Group

**•Pickup Group**

If this extension belongs to a pickup group, any calls that ring this extension can be picked up by other extensions in the same pickup group by dialing the Call Pickup feature code (default is\*4).

**Note:** \*4 is the default setting, it can be changed under Feature Codes -> General -> Call Pickup.

### 5) Call Duration Settings

Setup the max call duration for every call of this extension, but it's only valid for outbound calls. Enter '0' or leave this blank empty, the value would be equal to the max call duration configured in the Option Settings page.

**Note:** This setting will not be valid for internal calls.

### 6) VoIP Settings

#### •NAT

This setting should be used when the system is using a public IP address to communicate with devices hidden behind a NAT device (such as a broadband router). If you have one-way audio problems, you usually have problems with your NAT configuration or your firewall's support of SIP and/or RTP ports.

#### •Qualify

Send check alive packets to IP phones

#### •Enable SRTP

Enable extension for SRTP (RTP Encryption).

#### •Transport

This will be the transport method used by the extension. The options are UDP (default) or TCP or TLS.

•DTMF Mode – RFC2833, Info, Inband, Auto.

#### •Remote Register

Allow to register remote extensions.

This option is used to enhance the system security, it's disabled by default.

More details for the system security configuration please refer to **APPENDIX GMyPBX Security Configuration Guide**.

### 7) Other Options

#### •Call Waiting

Check this option if the extension should have Call Waiting capability. If this option is checked, the 'When busy' follow me options will not be available. The call waiting function of IP phone has higher priority than MyPBX's call waiting function.

#### •DND

Don't Disturb. When DND is enabled for an extension, the extension will be not available.

#### •User Web Interface

Check this option to allow the user to login to the MyPBX User Web interface,

which can be used to check voicemail and extension recordings. Users may login to MyPBX User Web interface by using their extension number and voicemail PIN # as the login and password respectively.

### **.Ring Out**

Check this option if you want to customize the ring time. Ring tone will stop over the time defined.

#### 8) Follow me (Call Forwarding)

Call forwarding for an extension can be configured here. The administrator can configure FollowMe option for this extension. If you want to transfer the call to an outbound number, please follow the dial pattern of outbound route filled in the outbound number.

For example: transferring to your mobile phone number 123456789, the dial pattern of outbound route is '9.', you should fill in 9123456789 here.

#### 9) IP Restriction

##### **-Enable IP Restriction**

Check this option to enhance the VoIP security for MyPBX. If this option is enabled, only the permitted IP/Subnet mask will be able to register this extension number. In this way, the VoIP security will be enhanced.

##### **-Permitted 'IP address/Subnet mask'**

The input format should be 'IP address'+'/'+'Subnet mask'.

e.g. "192.168.5.100/255.255.255.255" means only the device whose IP address is 192.168.5.100 is allowed to register this extension number.

e.g. "192.168.5.0/255.255.255.0" means only the device whose IP address is 192.168.5.XXX is allowed to register this extension number.

#### 10) Mobility Extension

MyPBX allows you to use your mobile phone as extension. If you set your mobile phone as mobility extension and then you call MyPBX with this mobility phone, you will hear a dial tone. MyPBX will recognize your call as a call from an extension. You can dial the number of other extensions (Your caller ID will be the number of your extension) or dial out via outbound routes just like dialing from your extension.

##### **-Mobility Extension Number**

Don't forget to add the dial patterns of the outbound route at the beginning of your mobile phone number when you fill in the mobility extension number filed. E.g. if you want to set "15960XXXXXX" as mobile extension, and the dial pattern of the outbound route is "9"; you should set "915960XXXXXX" here.

**Note:** If callback is enabled in the inbound route, the mobility extension function of this inbound route will be disabled

### 11) Spy Settings

MyPBX allows extension to monitor/barge in other conversation. Once this feature is enabled, the extension has the ability to monitor/barge in other calls using the feature codes for each spy mode, refer to 'Feature Codes' page for more information.

#### **•spy modes**

There are 4 spy modes available for choice:

General spy: you have the permission to use the following 3 modes.

Normal spy: you can only hear the call, but can't talk

Whisper spy: you can hear the call, and can talk with the monitored extension

Barge spy: you can hear the call and talk with them both

**Notes:** for example, if 500 want to monitor extension 501, we need to enable the 'allow being spied' for 501, and choose the spy mode for extension 500.

Then pick up 500 and dial "feature codes + 501" to start monitoring when 501 is in a call

If 500 chooses 'normal spy', it should dial '\*90501' to start monitoring;

If 500 chooses 'whisper spy', it should dial '\*91501' to start monitoring;

If 500 chooses 'barge spy', it should dial '\*92501' to start monitor;

If 500 chooses 'general spy', it can dial '\*90501', '\*91501' or '\*92501' to start monitor.



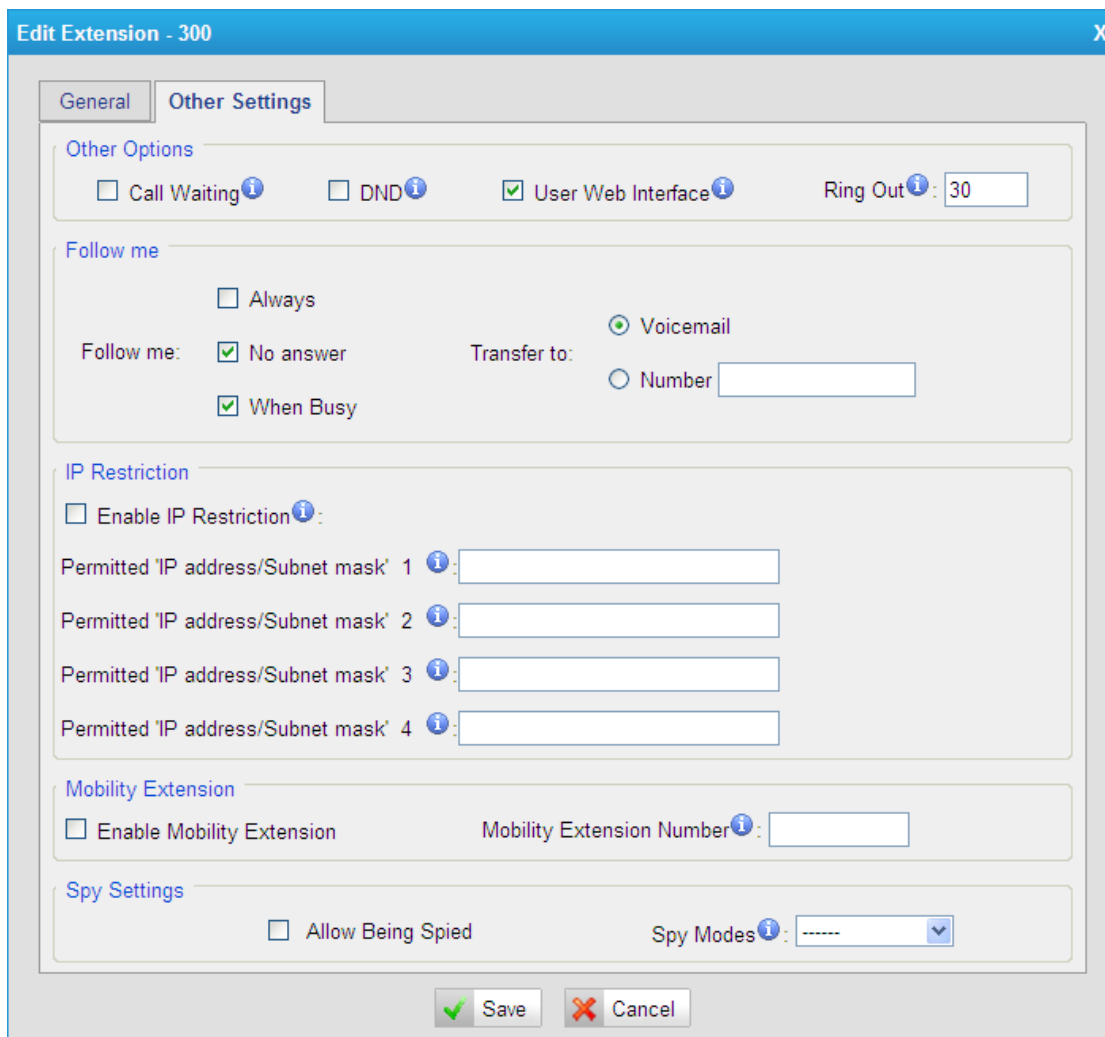


Figure 6-7 VOIP Extension Other Settings

## 6.1.2 Phone Provisioning

The Auto Provision sub menu provides users a method to Auto Provision IP Phone after the Express Setup process.

**Note:** Auto Provision functions fully test with these models:

**Yealink** (T12, T18, T20, T22, T26, T28, T32, T38, VP530, VP-2009)

**Snom** (300, 320, 360, 370)

**Snom** (300, 320, 360, 370)

**Polycom** (IP 6000, IP 7000, IP 32X, IP33X, IP430, IP450, IP550, IP560, VVX1500)

**Cisco** (IP7940, IP7960)

**Aastra**(9480i, 9480i-CT, 6730i, 6731i, 6753, 6755i, 6757i, 6757i CT, 6737i)

**GrandStream**(GXP1450, GXP2100, GXP2110, GXP2120)

**Escene**(ES220, ES320, ES330, ES410, ES610)

**Fanvil**(C56, C58, C60, C62)

**News:**

**When provisioning Yealink and Snom IP phone, MyPBX is not needed to be set as the only DHCP server any more.**

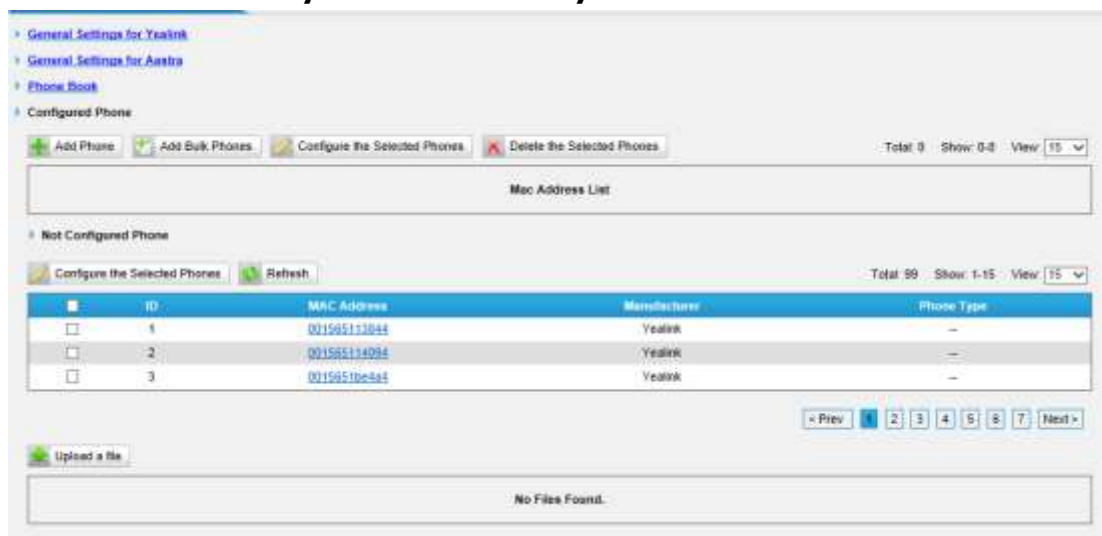


Figure 6-8 Phone Provisioning Page

### 6.1.2.1 General Settings for Yealink

In this page, you can configure it before provisioning Yealink IP phones, including the items like general preferences, codecs, remote phone book and firmware upgrade.

**Note:** if firmware download server is enabled, IP phone will update the firmware automatically according the version and server you have configured during the provision process.

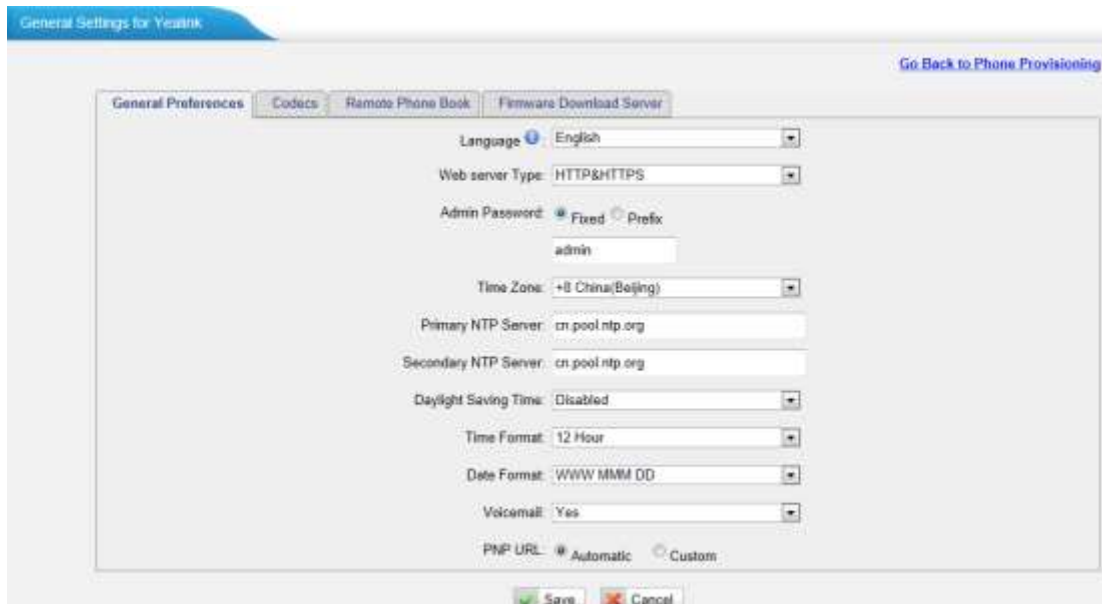


Figure 6-9 General Settings for Yealink

### 6.1.2.2 Aastra General Settings

In this page, you can configure it before provisioning Aastra IP phones,

including the items like general preferences, program keys configuration, soft keys configuration.

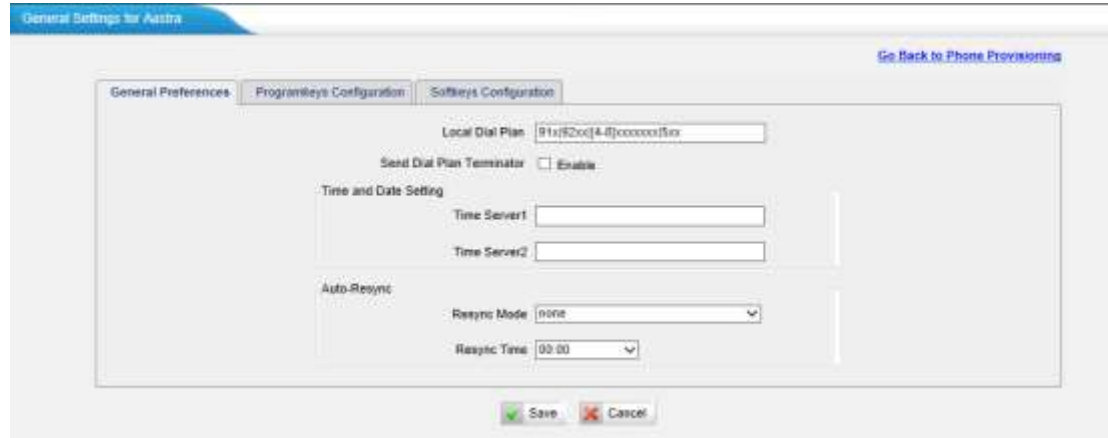


Figure 6-10 General Setting for Aastra

### 6.1.2.3 Phone book

You can add your contacts here and when you use phone provisioning, IP phone will download the phone book.

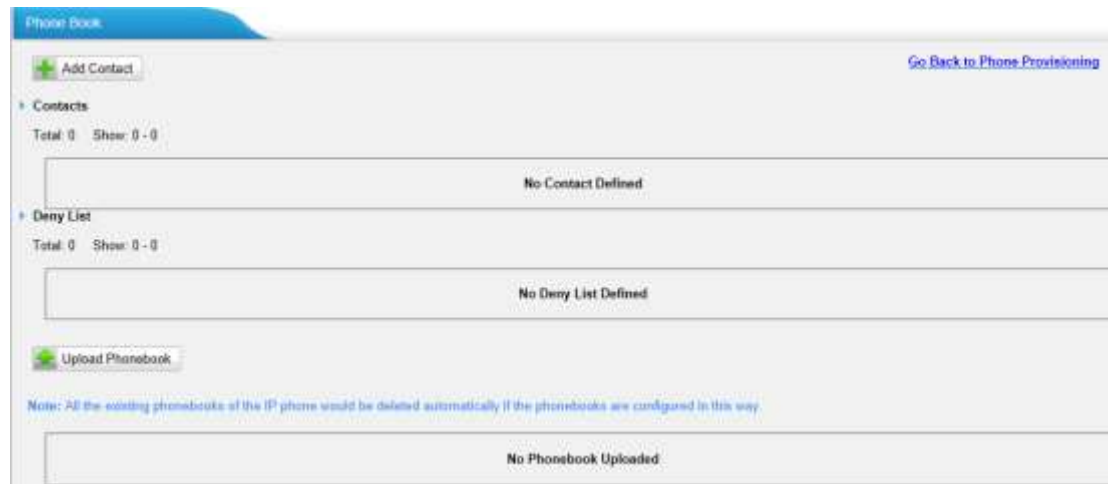


Figure 6-11 Phone Book

1) Add Contact

#### •Type

There are three types: None, VIP and Deny list (Blacklist).

#### •Group

There are 5 groups: None, Friends, Family, Work, Colleagues list.

#### •Nick Name

You can set a nick name for this number.

#### •Favorite

Only works with snom phone.

**•Organization**

Input the organization of this contact. Only works with snom phone.

**•Title**

Input the title of this contact. Only works with snom phone.

**•Email**

Input the email of this contact. Only works with snom phone.

**•Birthday**

Input the birthday of this contact. Only works with snom phone.

**•First Name**

Input the first name of this contact. Only works with snom phone.

**•Family Name**

Input the family of this contact. Only works with snom phone.

**•Office Number**

Input the office number here

**•Mobile Number**

Input the mobile number here

**•Home Number**

Input the home number here

**•Sub Number**

Add sub number of this contact. Only works with snom phone.

**•Note**

Take some note of this contact. Only works with snom phone.

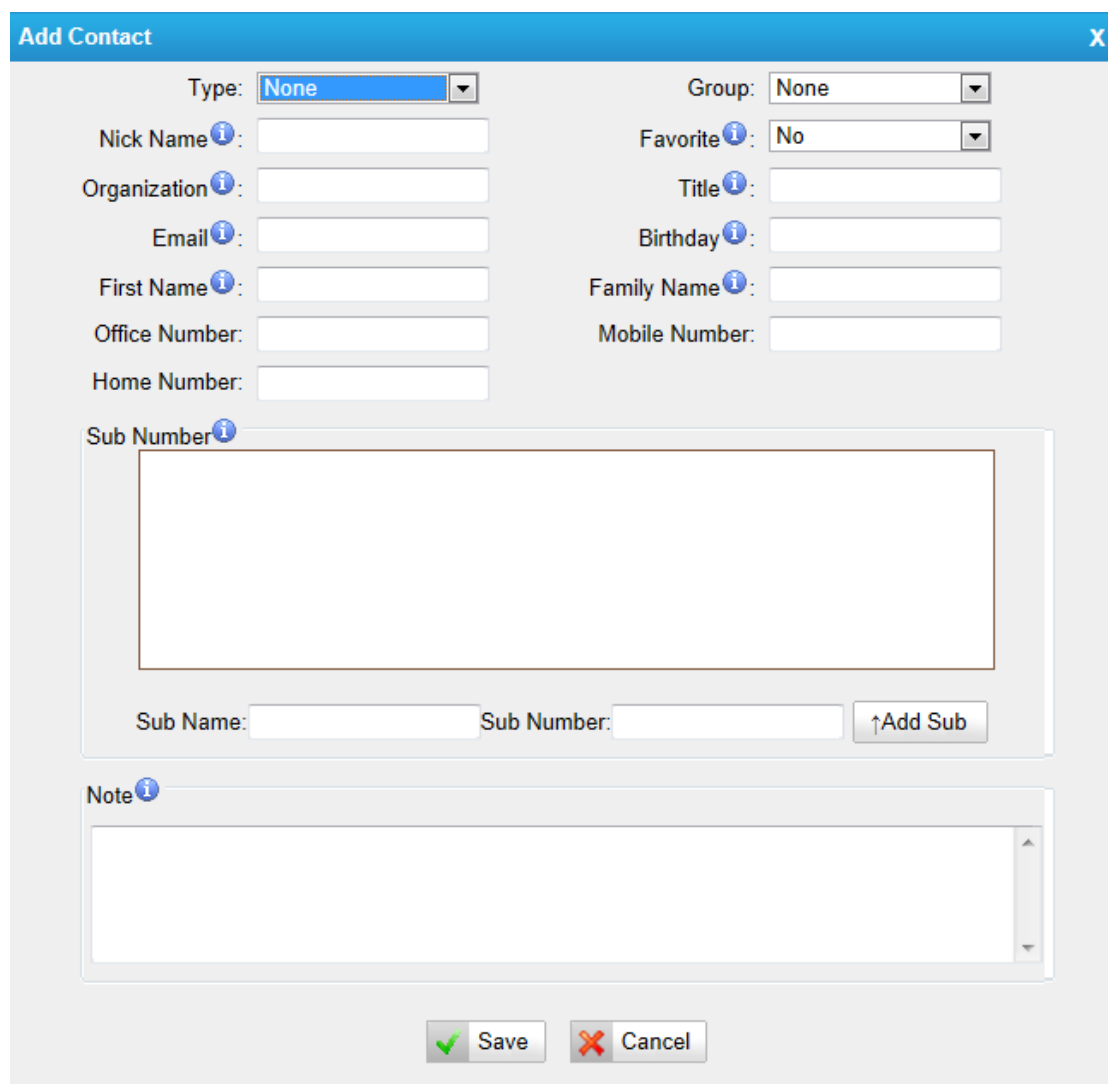


Figure 6-12 Add a Contact

## 2) Upload Phonebook

You can upload a phonebook before auto provision, which will be provisioned to the IP phone when using auto provision feature to configure your IP phones. The format of phonebook should be \*.xml.

**Note:** All the existing phonebooks of the IP phone will be replaced automatically if the phonebooks are configured in this way.

### 6.1.2.4 Configure phone

Let's take provisioning Yealink as an example.

Create New Phone have two modes,

Create New phone in webpage and Upload the IP Phone's configure file.

#### Add new phone via webpage

Click "Add Phone" and fill in the corresponding information in the pop-up window.

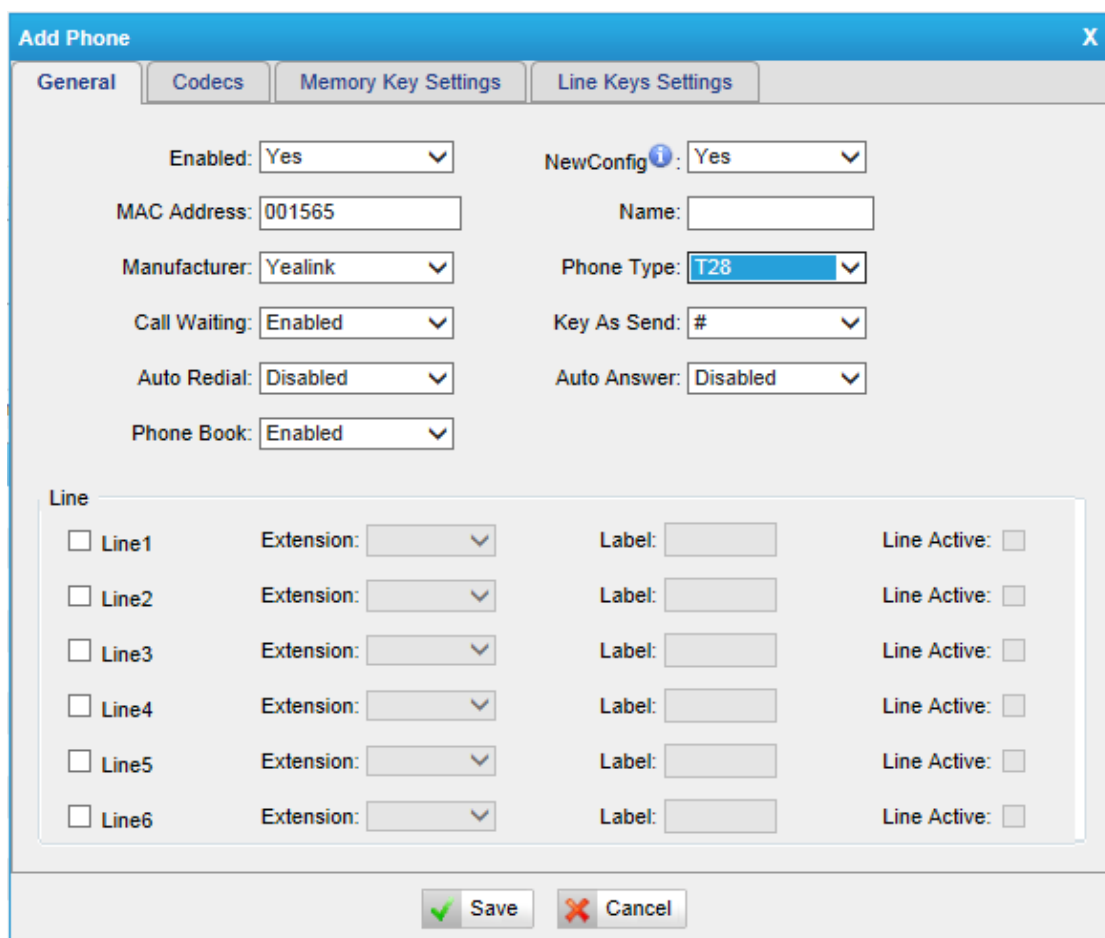


Figure 6-13 Configure Yealink T28

## 1) General

### •Enabled

Choose yes or no to enable or disable this extension

### •New Config

If your IP phone's firmware version is above x.70.x.x, you should select "Yes". Or else, it should be "No".

### • MAC address

Input the MAC address of IP phone

### •Name

Put the name of this Phone here.

### •Manufacturer

You can choose the Manufacturer of IP phone

### •Phone Type

Choose the model of your phone. Only for snom phone

### •Call Waiting

This call feature allows your phone to accept other incoming calls to an extension already in an active call.

### •Key as Send

Configure the key as send, you choose # ,\* or disable it

### •Auto redial

Enable the auto redial for IP Phone

### •Auto answer

Configure if auto answer is allowed for IP phone

### •Phone book

Enable the feature of phone book of IP phone

### •Line

You can set each line of IP phone for the account you want, active or not.

Extension: Selected the extension number for IP Phone.

Label: It is shown on the LCD for users to identify the account.

Line Active: You can choose on/off to enable/disable the account respectively.

## 2) Codecs

In this page, we can set the codecs for IP phone.

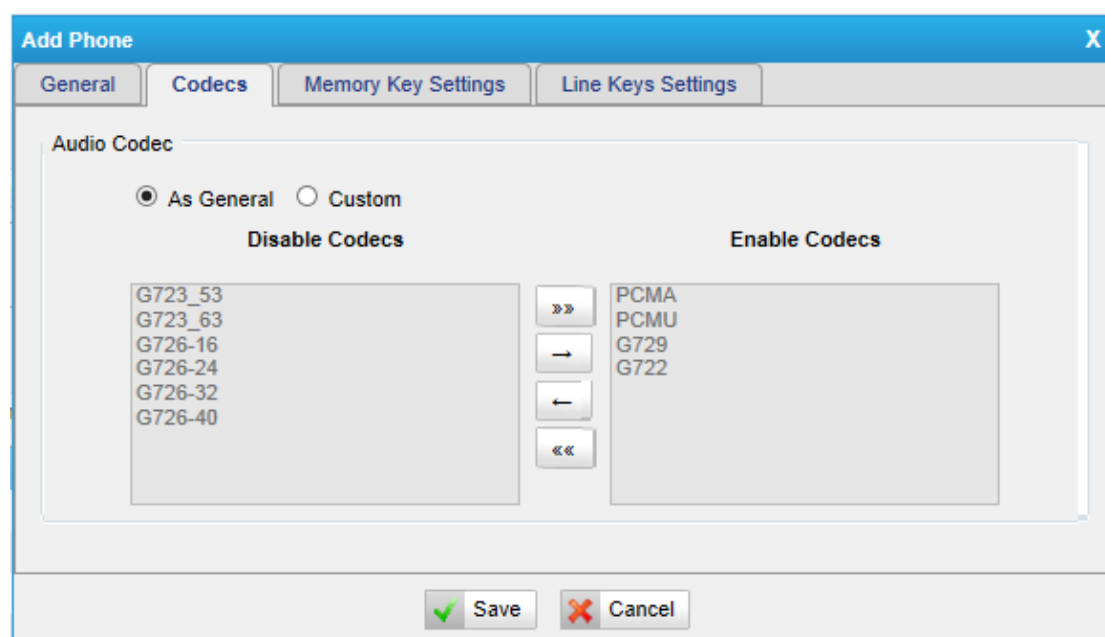
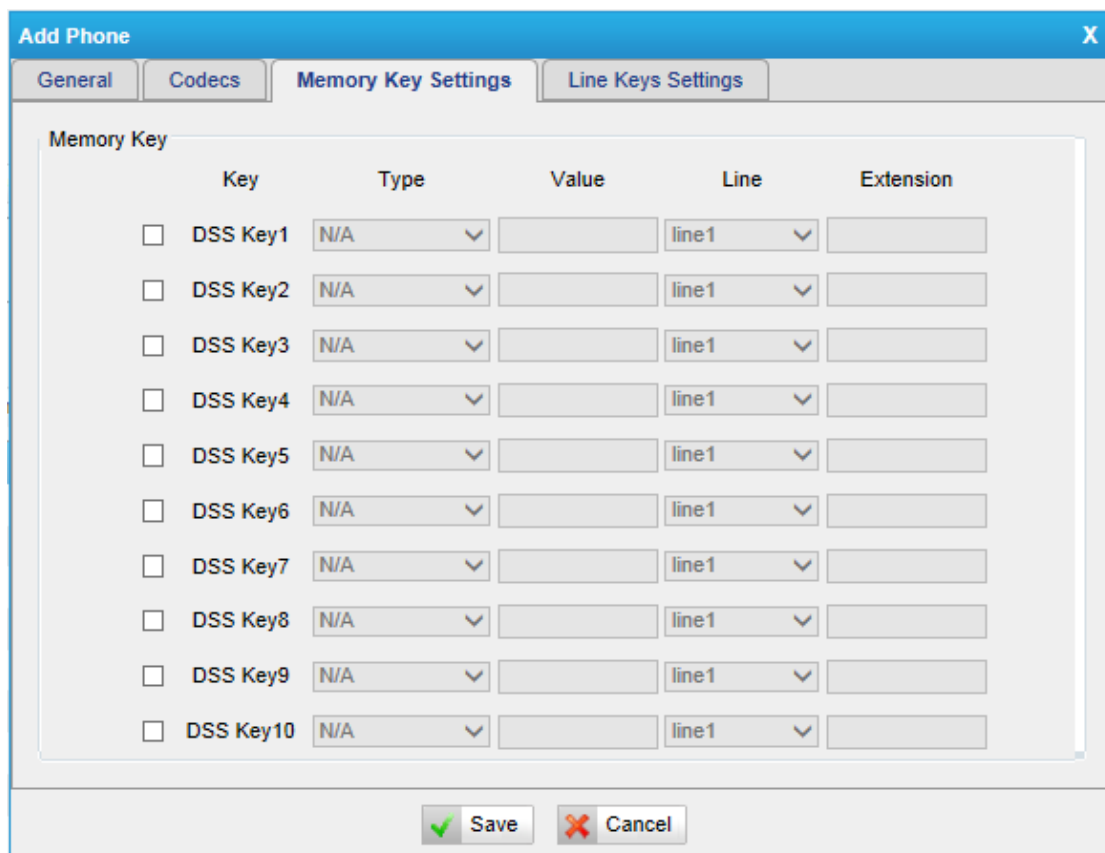


Figure 6-14 Select the Codec

### 3) Memory key settings

In this page, we can configure the DSS keys of IP phone one by one.

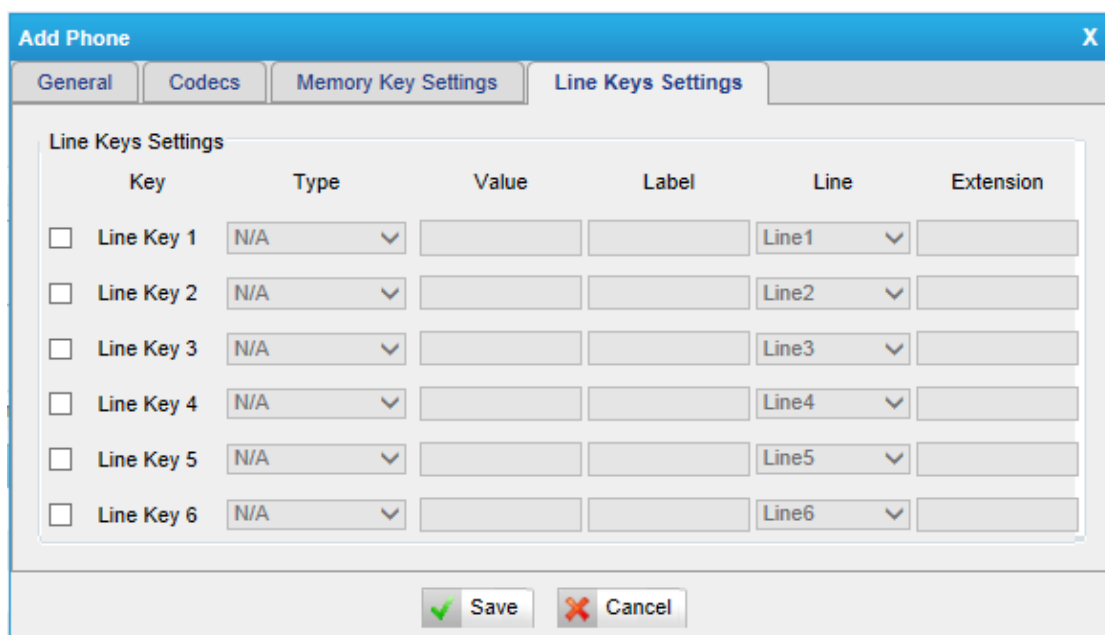


	Key	Type	Value	Line	Extension
<input type="checkbox"/>	DSS Key1	N/A		line1	
<input type="checkbox"/>	DSS Key2	N/A		line1	
<input type="checkbox"/>	DSS Key3	N/A		line1	
<input type="checkbox"/>	DSS Key4	N/A		line1	
<input type="checkbox"/>	DSS Key5	N/A		line1	
<input type="checkbox"/>	DSS Key6	N/A		line1	
<input type="checkbox"/>	DSS Key7	N/A		line1	
<input type="checkbox"/>	DSS Key8	N/A		line1	
<input type="checkbox"/>	DSS Key9	N/A		line1	
<input type="checkbox"/>	DSS Key10	N/A		line1	

Figure 6-15 Configure Memory Key

### 4) Line keys settings

We can configure the line key settings for this IP phone



	Key	Type	Value	Label	Line	Extension
<input type="checkbox"/>	Line Key 1	N/A			Line1	
<input type="checkbox"/>	Line Key 2	N/A			Line2	
<input type="checkbox"/>	Line Key 3	N/A			Line3	
<input type="checkbox"/>	Line Key 4	N/A			Line4	
<input type="checkbox"/>	Line Key 5	N/A			Line5	
<input type="checkbox"/>	Line Key 6	N/A			Line6	

Figure 6-16 Configure Line Key



#### 6.1.2.4 Not configured phone

In this section, MyPBX will scan all the supported IP phones and display here, we can click the 'MAC address' of IP phone and input the corresponding information in the pop-up window, like figure 6-13.



ID	MAC Address	Manufacturer	Phone Type
1	0015651208d5	Yealink	---
2	001565148165	Yealink	---
3	0015651118d0	Yealink	---
4	0015652-2c00	Yealink	---
5	0015651118d0	Yealink	---
6	001565299102	Yealink	T20

Figure 6-17 Not Configured Phone List

#### 6.1.2.5 Upload a file

Click "Upload a file" and choose the configure file of IP phone in the popup window.

**Note:** the file format must be

Yealink: .cfg file

Snom: .htm file

Grandstream: .xml file

Please edit the configuration files in advance before uploading.

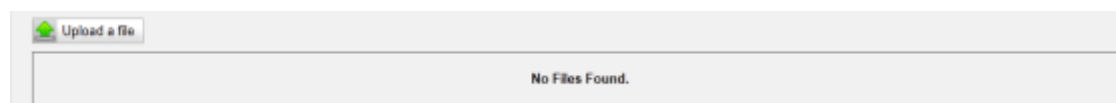


Figure 6-18 Upload Configuration File

## 6.2 Trunks

There are two types in MyPBX U300, the VOIP trunk and the E1 trunk. We cannot use other trunks like PSTN/GSM/BRI with MyPBX U300.

### 6.2.1 Physical Trunk

In this page we can configure the details of E1 trunk, before configure anything, please make sure the cable is fine, and you have got enough information from the ISDN provider.



Figure 6-19 Physical Trunk-E1

On the Physical Trunk page, click 'Edit' on the selected trunk and modify its properties in the popup window:

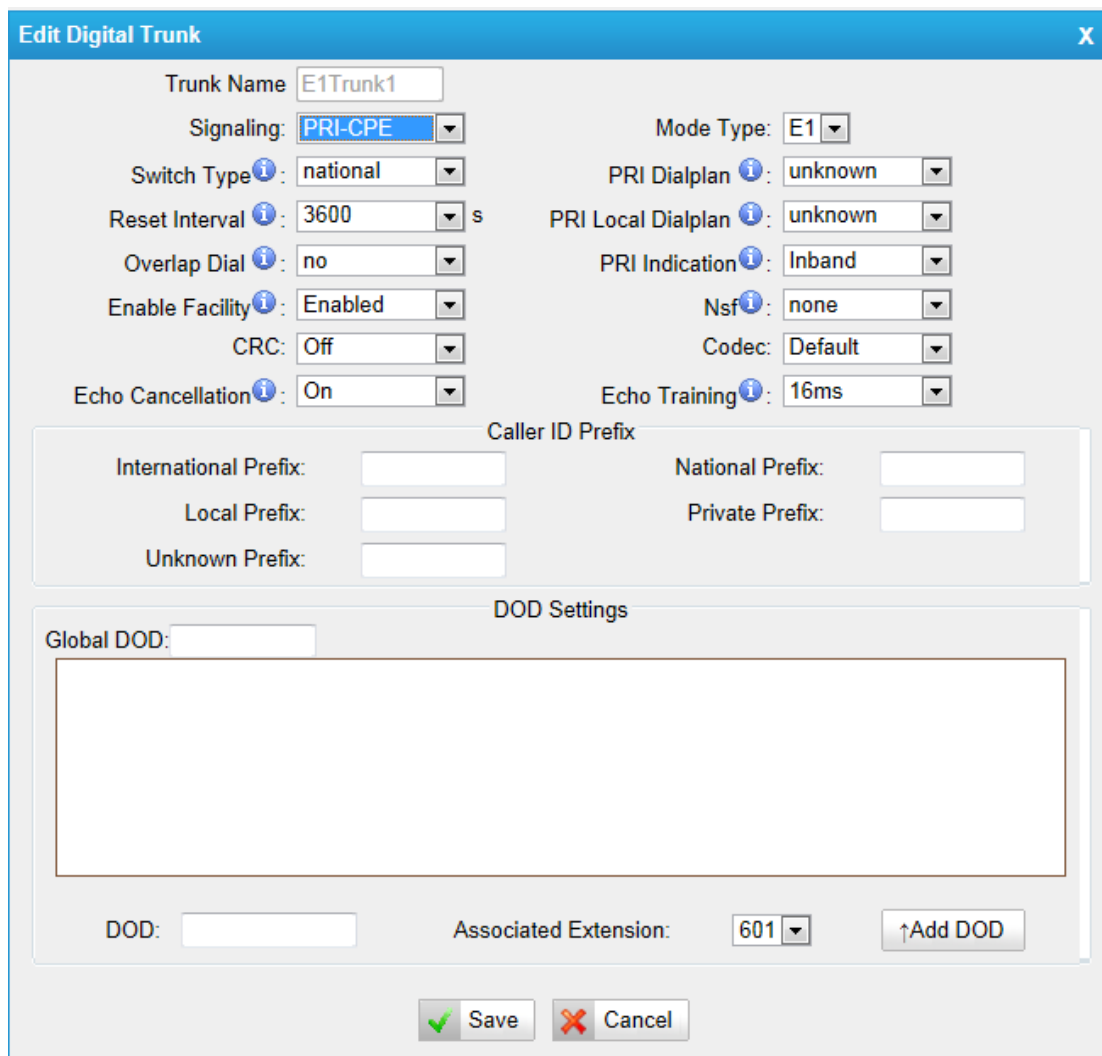


Figure 6-20 Edit E1 Trunk

#### • Trunk name

Set the trunk name

#### •Signaling

PRI-NET, PRI-CPE,MFC/R2,SS7

#### •Mode Type

E1 / T1 / J1

#### •Switch Type

national: National ISDN type2 (common in the US)

ni1: National ISDN type 1

dms100: Nortel DMS100

4ess: AT&T 4ESS

5ess: Lucent 5ESS

euroisdn: EuroISDN

qsig: Minimalistic protocol to build a "network" with two or more PBX of different

vendors!

**•PRI Dial Plan**

Sets an option required for some (rare) switches that require a dial plan parameter to be passed. This option is ignored by most PRI switches. It may be necessary on a few pieces of hardware. This option can almost always be left unchanged from the default.

**•Reset interval**

Sets the time in seconds between restart of unused channels. Some PBXs don't like channel restarts. so set the interval to a very long interval e.g. 100000000 or 'never' to disable \*entirely\*. If you are in Israel, the following is important: As Bezeq in Israel doesn't like the B-Channel resets happening on the lines, it is best to set the reset interval to 'never' when installing a box in Israel. Our past experience also shows that this parameter may also cause issues on local switches in the UK and China.

**•PRI Local Dial Plan**

Sets an option required for some (rare) switches that require a dial plan parameter to be passed. This option is ignored by most BRI switches. It may be necessary on a few pieces of hardware. This option can almost always be left unchanged from the default.

**•Over Lap Dial**

Define whether MyPBX can dial this switch using overlap digits. If you need Direct Dial-in (DDI; in German "Durchwahl") you should change this to yes, then MyPBX will wait after the last digit it receives.

**•PRI Indication**

Tells how Device should indicate Busy() and Congestion() to the switch/user. Accepted values are:

inband: Device plays indication tones without answering; not available on all PRI/BRI subscription lines .

outofband: Device disconnects with busy/congestion information code so the switch will play the indication tones to the caller. Busy() will now do same as setting PRI\_CAUSE=17 and Hangup().

**•Facility Enable**

To enable transmission of facility-based ISDN supplementary services (such as caller name from CPE over facility).

**•NSF**

Used with AT&T PRIs.If outbound calls are being rejected due to "Mandatory information element missing" and the missing IE is 0x20, then you need this

setting.

**•CRC**

CRC Verification.

**•Codec**

Default,A-Law,U-Law.

**•Echo Cancellation**

Obviously this disables or enables echo cancellation, it is recommended not to turn this off.

## **1) CallerID Prefix**

**•International Prefix**

When there are international calls coming in via this BRI trunk, the International Prefix you have set here will be added before the CID. So you can know this is an international call before you answer it.

**•National Prefix**

When there are national calls coming in via this BRI trunk, the National Prefix you have set here will be added before the CID. So you can know this is a national call before you answer it.

**•Local Prefix**

When there are Local calls coming in via this BRI trunk, the Local Prefix you have set here will be added before the CID. So you can know this is a local call before you answer it.

**•Private Prefix**

When there are Private calls coming in via this BRI trunk, the Private Prefix you have set here will be added before the CID. So you can know this is a Private call before you answer it.

**•Unknown Prefix**

When there are calls with unknown number coming via this BRI trunk, the Unknown Prefix you set here will be shown as the caller ID.

## **2) DOD Setting**

DOD (Direct Outward Dialing) means the caller ID displayed when dialing out, before configure this, please make sure the provider supports this feature

**•Global DOD**

Global direct outward dialing number.

### •DOD

Direct Outward Dialing Number.

### •Associated Extension

The extension make call out via BRI Trunk will display the associated DOD.

## 6.2.2 VoIP Trunk

There are two types of VOIP trunk in MyPBX: SIP and IAX, in this page, we can also configure the 'service provider' trunk, which doesn't need the use name and password for authorization, when you have bought a trunk from provide with IP address only, please choose 'service provider' trunk .



Figure 6-21 VOIP Trunk

### 6.2.2.1 VoIP Trunk

In this page, we can configure VoIP trunk (SIP/ IAX) you have got from provider with the authorization name and password.

#### 1) Add SIP Trunk

Input correct SIP information (provide by VoIP provider). Inaccurate information will prevent the trunk from registering.

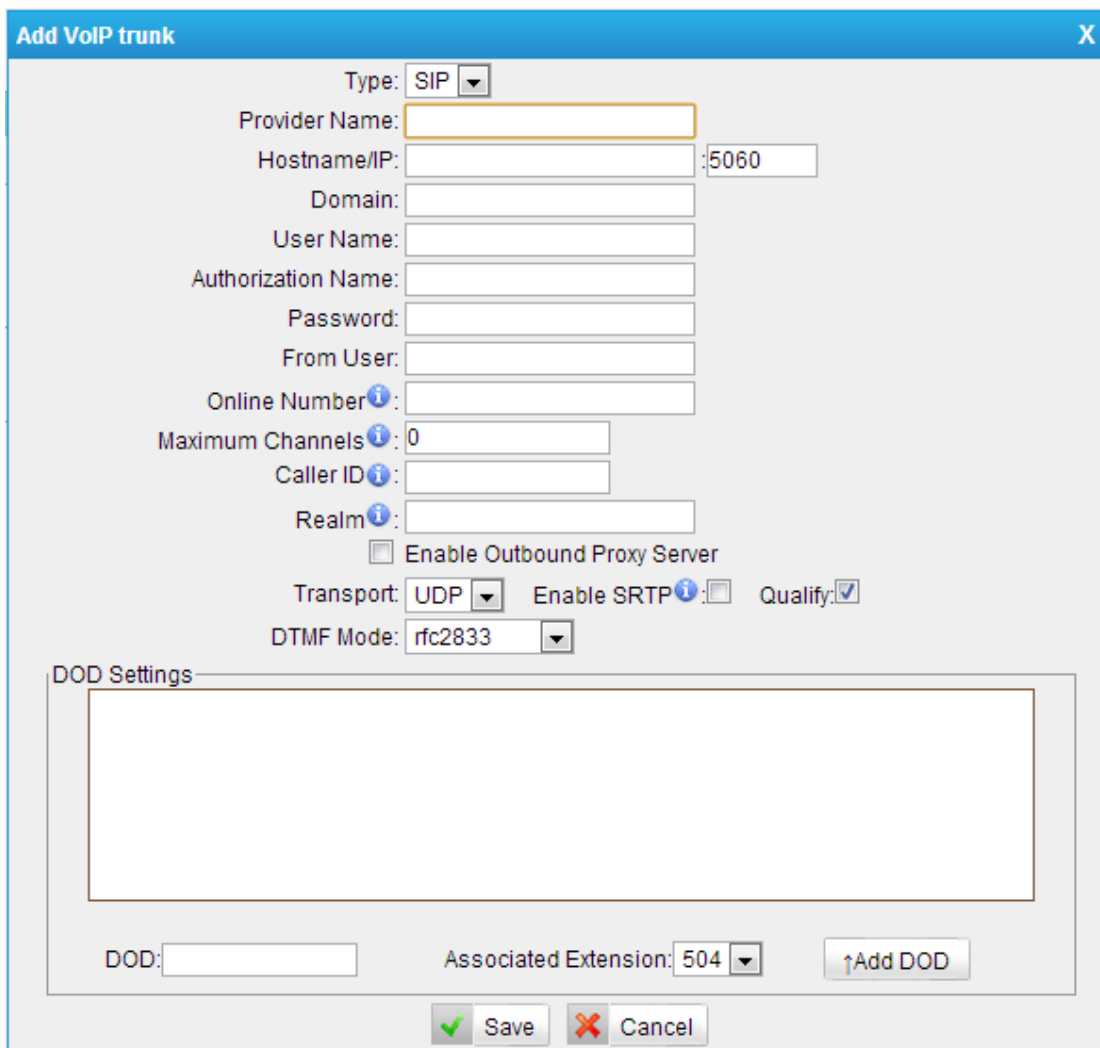


Figure 6-22 Add a SIP Trunk

### •Type

SIP – Identifies whether the trunk sends and receives calls using the VoIP protocol SIP

### •Provider Name

A unique label to help you identify this trunk when listed in outbound rules, incoming rules etc. E.g. "yeastar".

### •Hostname/IP

Service provider's hostname or IP address. 5060 is the standard port number used by SIP protocol. Don't change this part if it is not required.

### •Domain

VoIP provider's server domain name.

### •Username

Username of SIP account. Used for SIP trunk registration.

**.Authorization name**

Used for SIP authentication. Leave this blank if not required.

**.Password**

Password of SIP account.

**.From User**

All outgoing calls from this SIP Trunk will use the From User (In this case the account name for SIP Registration) in From Header of the SIP Invite package. Keep this field blank if not needed

**.Online number**

Define the online number that expected by 'Skype Connect' and some other SIP service providers. Leave this field blank if not needed.

**.Maximum Channels**

Controls the maximum number of outbound channels (simultaneous calls) that can be used on this trunk . Inbound calls are not counted against the maximum. Set as 0 to specify no maximum.

**.Caller ID**

Specify the caller ID to use when making outbound calls over this trunk. The caller ID set in the 'extension' screen will override the caller ID set in the 'VOIP trunk' screen. Please note that not all the service providers support this feature. Contact your service provider for more information.

**.Realm**

Realm is a string to be displayed to users so they know which username and password to use.

**.Outbound Proxy Server**

A proxy that receives requests from a client, even though it may not be the server resolved by the Request-URI.

**.Codecs**

Define the codec for this sip trunk and its priority

**Note:** To change the codec type and priority of this trunk, please create it first, it will appear when you edit it again.

**.Transport**

This will be the transport method used by the SIP Trunk. This method is given by the SIP trunk provider. The options are UDP (default) or TCP or TLS.



### •Enable SRTP

Define if SRTP is enabled for this trunk

### •Qualify

Send check alive packets to the sip provider.

### •DTMF mode

Set default mode for sending DTMF of this trunk. Default setting: rfc2833

### •DOD

DOD (Direct Outward Dialing) means the caller ID displayed when dialing out, before configure this, please make sure the provider supports this feature

### •Associated Extension

The extension make call out via SIP Trunk will display the associated DOD

## 2) Add IAX trunk

Input correct IAX information (provided by VOIP provider). Inaccurate information will prevent the trunk from registering.

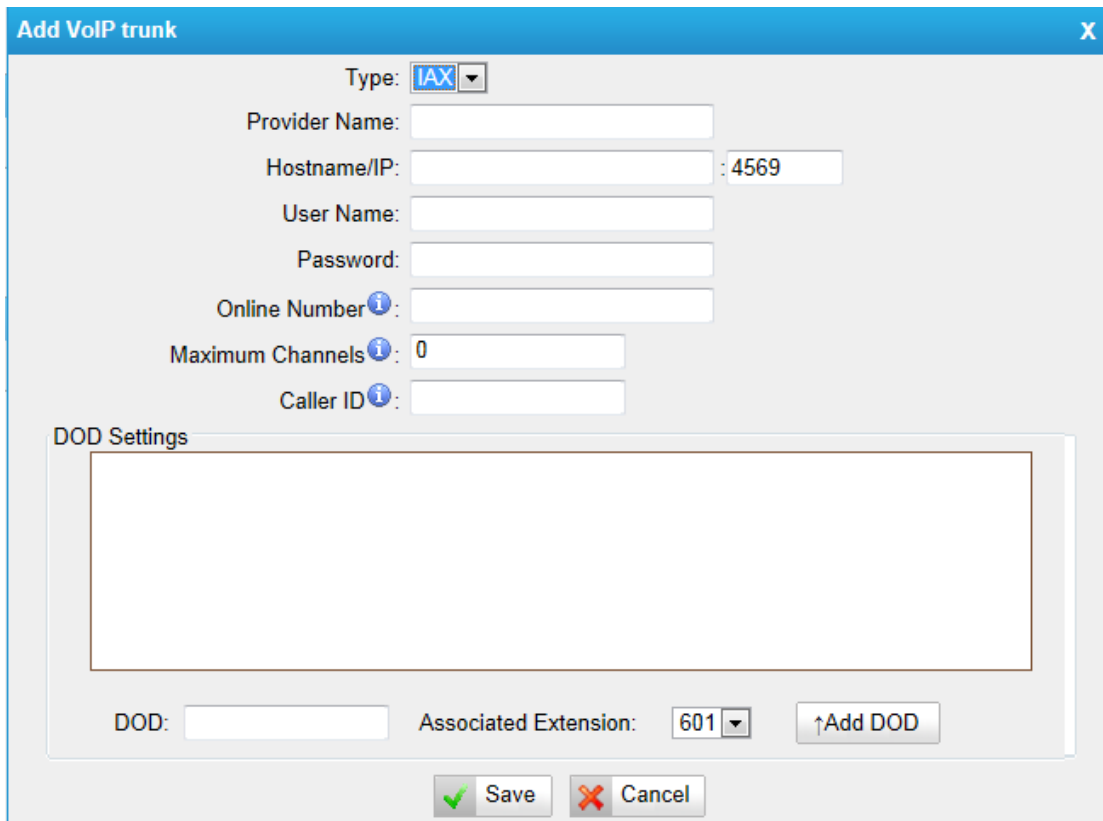


Figure 6-23 Add an IAX Trunk

### •Type

IAX – Identifies whether the trunk sends and receives calls by using the VoIP protocol IAX.

**•Provider Name**

A unique label to help you identify this trunk when listed in outbound rules, incoming rules etc. E.g. "yeastar2".

**•Hostname/IP**

Service provider's hostname or IP address. 4569 is the standard port number used by IAX protocol. Don't change this part if it is not required.

**•Username**

Username of IAX account. Used for IAX trunk registration.

**•Password**

Password of IAX account.

**•Online number**

Define the online number that expected by 'Skype Connect' and some other SIP service providers. Leave this field blank if it's no required.

**•Maximum Channels**

Control the maximum number of outbound channels (simultaneous calls) that can be used on this trunk . Inbound calls are not counted against the maximum. Set as 0 to specify no maximum.

**•Caller ID**

Specify the caller ID to use when making outbound calls over this trunk. The caller ID set in the 'extension' screen will override the caller ID setting in the 'VOIP trunk' screen. Please note that not all the service providers support this feature. Contact your service provider for more information.

**•DOD**

DOD (Direct Outward Dialing) means the caller ID displayed when dialing out, before configure this, please make sure the provider supports this feature

**•Associated Extension**

The extension make call out via IAX Trunk will display the associated DOD

**6.2.2.2 Service Provider**

This is service provider trunk (peer to peer mode), which authorized using IP address only. If you have got a trunk with IP address only, please choose this type

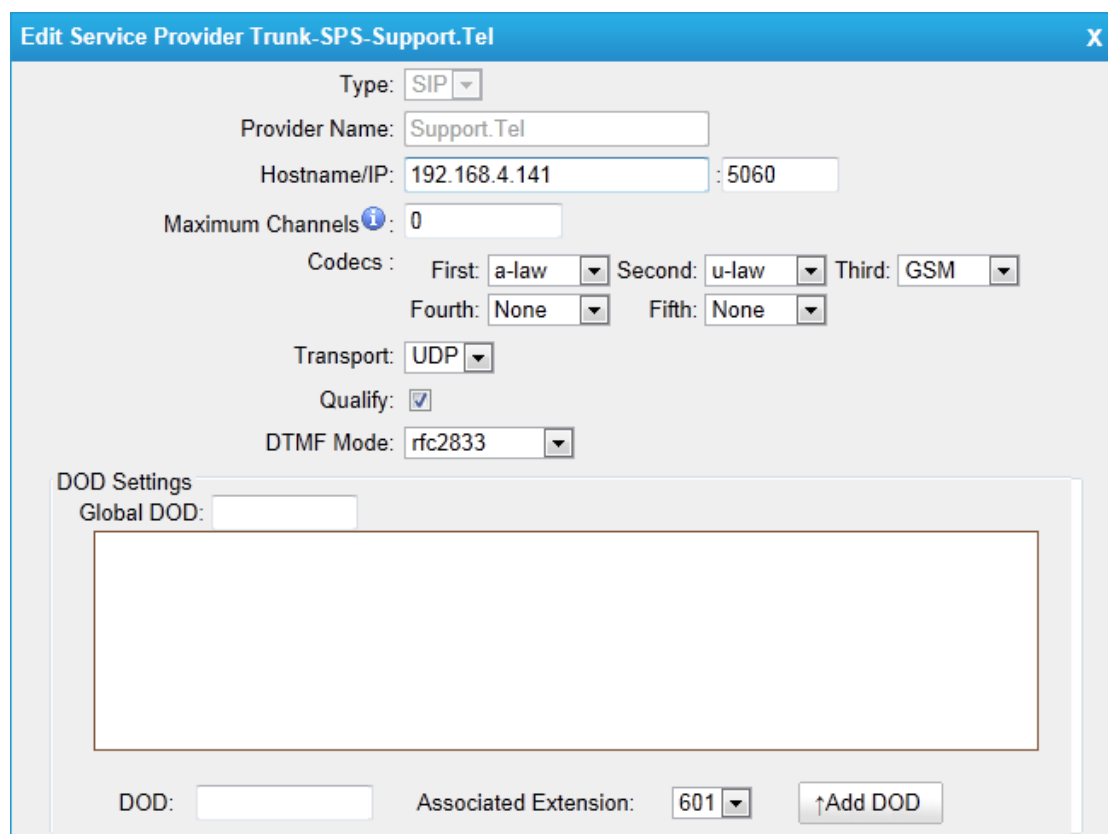


Figure 6-24 Service Provider Trunk

#### •Type

SIP or IAX

SIP – Identifies whether the trunk sends and receives calls by using the VoIP protocol SIP.

IAX - Identifies whether the trunk sends and receives calls by using the VoIP protocol IAX.

#### •Provider Name

A unique label would help to you identify this trunk. Ex: 'Provider2'.

#### •Hostname/IP

Service provider's hostname or IP address.

**Note:** 5060 is the standard port number used by SIP protocol, 4569 is the standard port number used by IAX protocol. Don't change this part if it is not required.

#### •Maximum Channels

Controls the maximum number of outbound channels (simultaneous calls) that can be used on this trunk. Inbound calls are not counted against the maximum. Leave blank to specify no maximum.

#### •Codecs

Define the codec for this sip trunk and its priority

**Note:** codec can only display when edit it after creating the trunk.

**•Transport**

This will be the transport method used by the SIP Trunk. This method is given by the SIP trunk provider. The options are UDP (default) or TCP or TLS.

**•Qualify**

Send check alive packets to the sip provider.

**•DTMF mode**

Set default mode for sending DTMF of this trunk. Default setting: rfc2833

**•DOD**

DOD (Direct Outward Dialing) means the caller ID displayed when dialing out, before configure this, please make sure the provider supports this feature

**•Associated Extension**

The extension make call out via this Trunk will display the associated DOD.

## 6.3 Outbound Call Control

### 6.3.1 Outbound Routes

In this page, we can configure the outbound rules to control the outgoing calls.

**Notes:**

1. The max number of outbound route is 64
2. If the dial patterns are the same in several routes, MyPBX will choose the available routes from top to the last one.
3. When you have created a new extension, please edit the outbound route so that he can dial out too

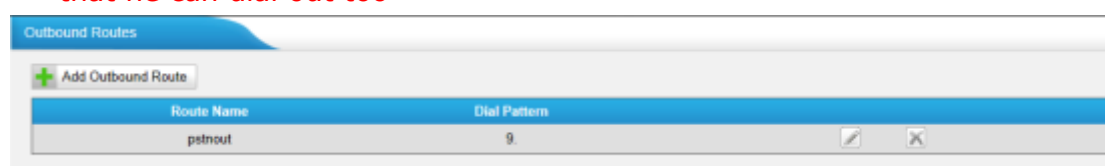


Figure 6-25 Outbound Route Page

We can create outbound route or use the default route 'pstnout' ( dial 9+numbers to dial out)

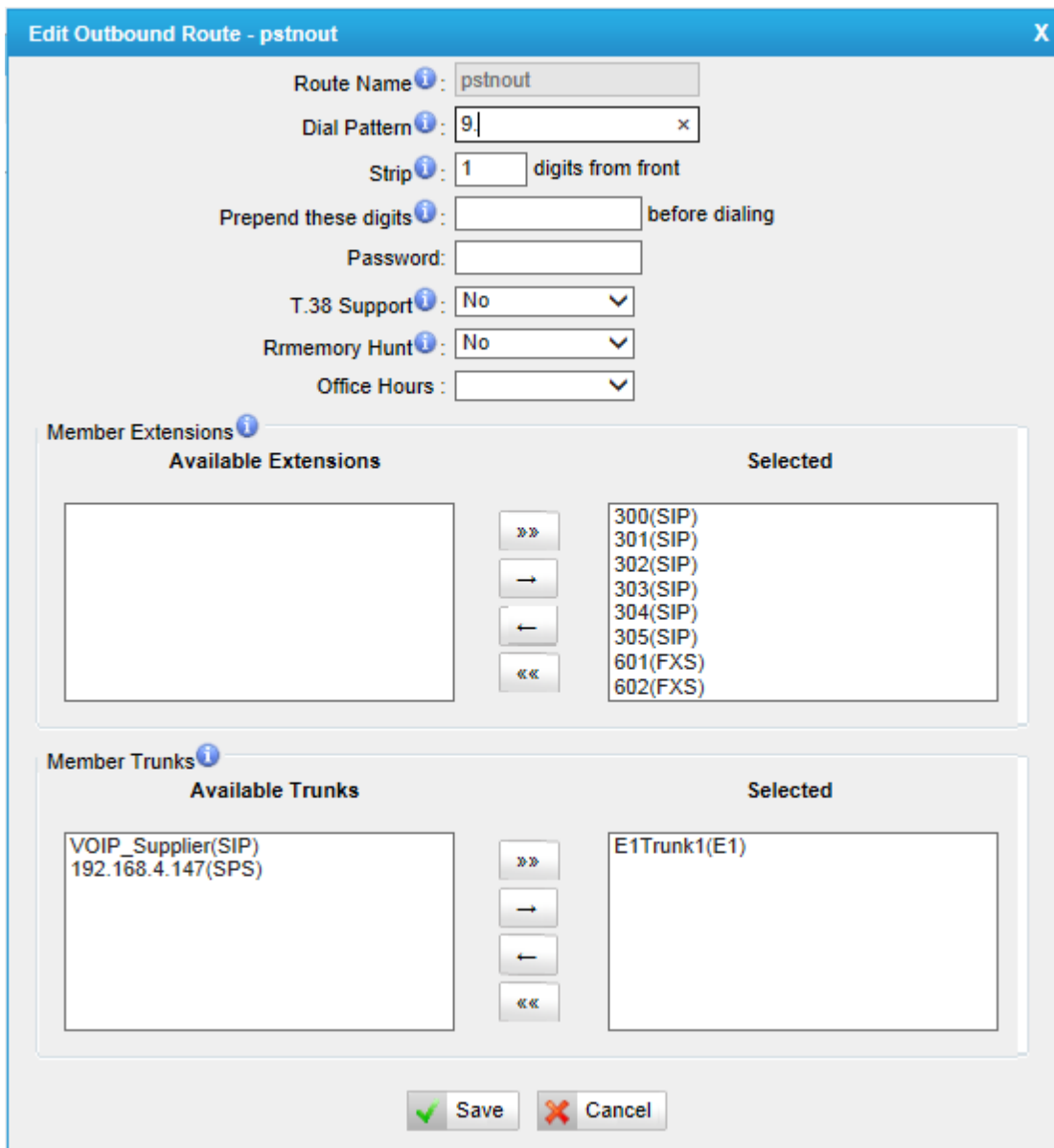


Figure 6-26 Add/Edit Outbound Route

### •Route Name

Name of this Outbound Route. Ex: 'Local' or 'Long Distance' etc.

### •Dial Pattern

Outbound calls that match this dial pattern will use this outbound route. There are a number of dial pattern characters that have special meanings:

**X**: Any Digit from 0-9

**Z**: Any Digit from 1-9

**N**: Any Digit from 2-9

**[12345-9]**: Any digit in the brackets (in this example, 1,2,3,4,5,6,7,8,9)

The '.' Character will match any remaining digits. For example, 9011. will match any phone number that starts with 9011, excluding 9011 itself.

The '!' will match none remaining digits, and causes the matching process to complete as soon as it can be determined that no other matches are possible.

Example 1: **NXXXXXX** will match any 7 digits phone number.

Example 2: **1NXXNXXXXXX** will match a phone number starting with a 1, followed by a 3-digit area code, and then 6 digit number.

**•Strip digits from front**

Allows the user to specify the number of digits that will be stripped from the front of the phone number before the call is placed. For example, if users must press 0 before dialing a phone number, one digit should be stripped from the dial string before the call is placed.

**•Prepend these digits before dialing**

These digits will be prepended to the phone number before the call is placed. For example, if a trunk requires 10 digit dialing, but users are more comfortable with 7 digit dialing, this field could be used to prepend a 3 digit area code to all 7 digit phone numbers before calls are placed.

**•Password**

The route password can be used to protect this route from being accessed without a password.

**•T.38 Support:**

Enable T38 fax in this outbound route(Only for SIP Trunk).

**•Rrmemory Hunt**

Round robin with memory, remembers which trunk was used last time, and then use the next available trunk to call out.

**•Office Hours**

When a specific office hour is selected, this outbound route can only be used during this office hour, and can't be used in non-office hours.

**•Member Extensions**

Defines the extensions that will be permitted to use this outbound route.

**•Member Trunks**

Define the trunks that can be used for this outbound route.

## 6.3.2 Speed Dial Settings



Figure 6-27 Speed Dial Page

### 1) Options

#### •The prefix of speed dial

The prefix should be dialed before the speed dial number. Default is \*99

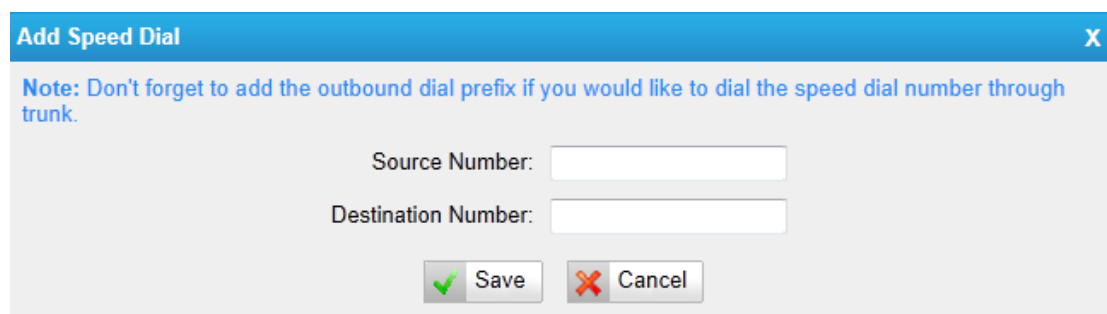


Figure 6-28 Speed Dial Settings

### 2) Add new speed dial.

#### •Source Number

The speed dial number.

#### •Destination Number

The number you want to call.

e.g. The source number is "123". The destination number is 5503305. The prefix number is \*99. You can use an extension with any type to dial \*99123, then it will call to number 5503305.

**Note:** Don't forget to add the outbound dial prefix if you would like to dial the speed dial number through trunk.



## 6.4 Inbound Call Control

In this page, we can configure the details of IVR, ring group, queue and inbound routes.

### 6.4.1 IVR

When there's an inbound call aims at Auto Attendant, MyPBX will play an IVR recording and route the caller to the requested destination (for example, 'Welcome to XX company, for sales press 1, for technical support press 2, for operator press 0, etc). The system will transfer the call to corresponding extension according to DTMF digits inputted by the user

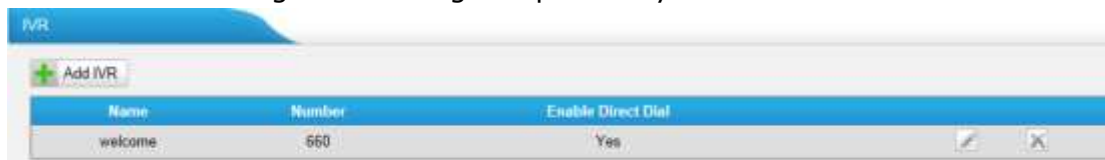
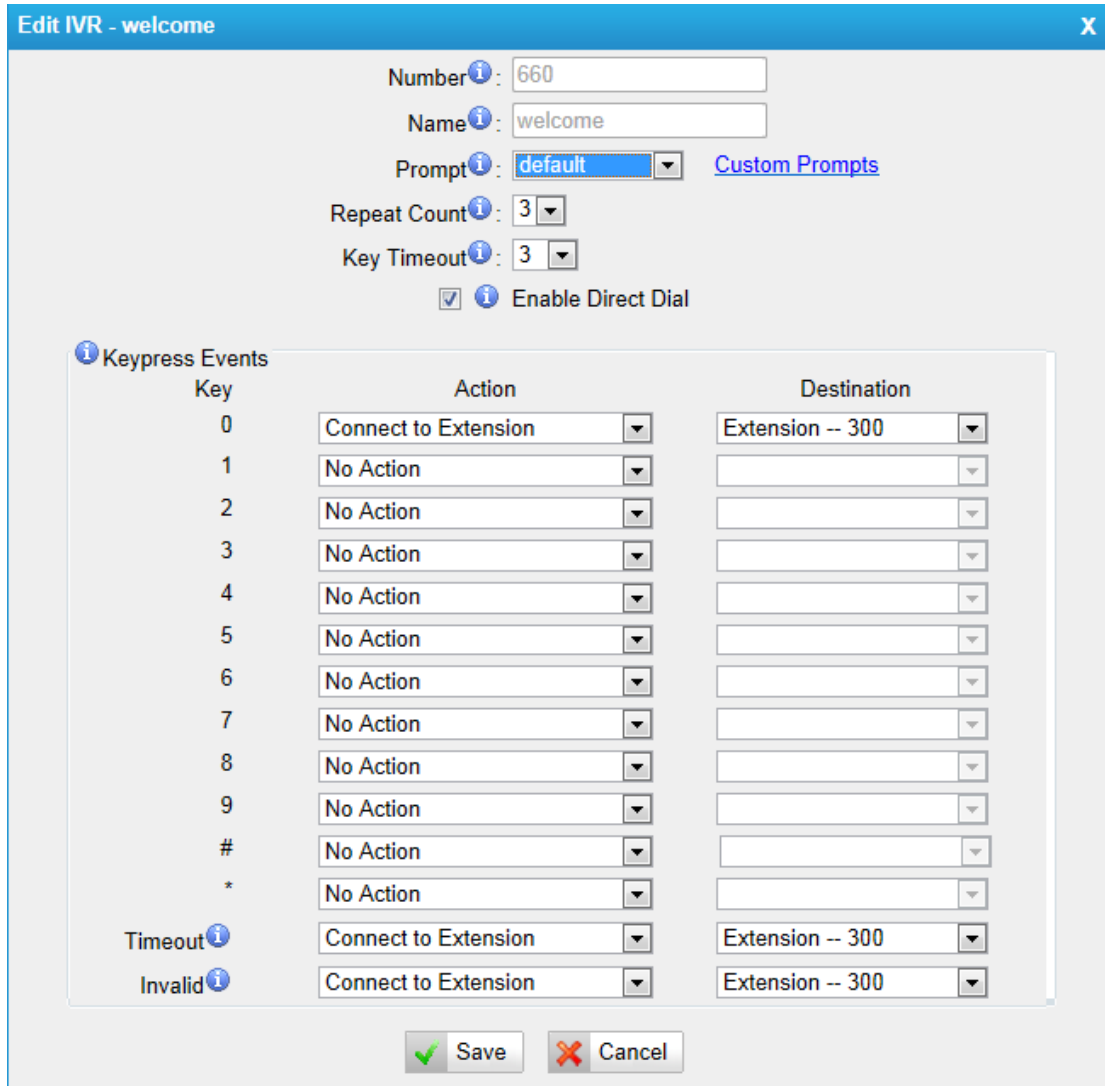


Figure 6-29 IVR Page

There is a default IVR here, we can edit it directly or add IVR by yourself



Number: 660

Name: welcome

Prompt: default [Custom Prompts](#)

Repeat Count: 3

Key Timeout: 3

☒ Enable Direct Dial

Key	Action	Destination
0	Connect to Extension	Extension -- 300
1	No Action	
2	No Action	
3	No Action	
4	No Action	
5	No Action	
6	No Action	
7	No Action	
8	No Action	
9	No Action	
#	No Action	
*	No Action	
Timeout	Connect to Extension	Extension -- 300
Invalid	Connect to Extension	Extension -- 300

Save Cancel

Figure 6-30 IVR Settings

#### •Number

MyPBX treats IVR as an extension; you can dial this extension number to reach the IVR from internal extension.

#### •Name

A name for the IVR

#### •Prompt

The prompt recording that will be played when this IVR is reached.

#### •Repeat Count

The number of times that the selected IVR prompt will be played.

#### •Key Timeout

Wait for the user to enter a new extension for a specified number of seconds.

**•Enable Direct Dial**

Allow the caller to dial other extensions number directly.

**•Key Press Events**

A list of actions that can be performed depending on the digit dialed by the user.

**•Key**

The Key pressed when the callers hear the IVR prompt.

**•Action**

When the callers press the corresponding key, the action MyPBX executes.

No Action: Do nothing

Connect to Extension: Connect the call to an extension.

Connect to Voicemail: Connect the call to the voicemail of an extension

Connect to RingGroup: Connect the call to a ringgroup.

Connect to IVR: Connect the call to an IVR.

Connect to Conference Room: Connect the call to a conference room.

Connect to DISA: Connect the call to a DISA.

Connect to Queue: Connect the call to a queue.

Connect to Faxes: Connect the call to Faxes of extensions.

Dial by Name: The callers can dial the name of an extension to connect to the corresponding extension.

Hung up: Hang up the call.

**•Destination**

Where will MyPBX route the call when the action occurs.

**•Time Out**

Defines the timeout action. A timeout occurs after the IVR prompt has finished playing for the number of times specified by the 'Repeat Count' field.

**•Invalid**

Defines the invalid action. The invalid action is triggered if the user enters a DTMF digit that is not defined for this IVR.

## 6.4.2 Ring Groups

Ring groups can be configured to balance the call traffic for multiple users and give callers a higher level of availability for incoming calls. Multiple ring methods and voicemail are supported.

**Note:** follow me feature in extension page will not take effect when it's ringing as an agent.



Figure 6-31 Ringgroup Page

There is a default ringgroup, you can edit it or create a new one

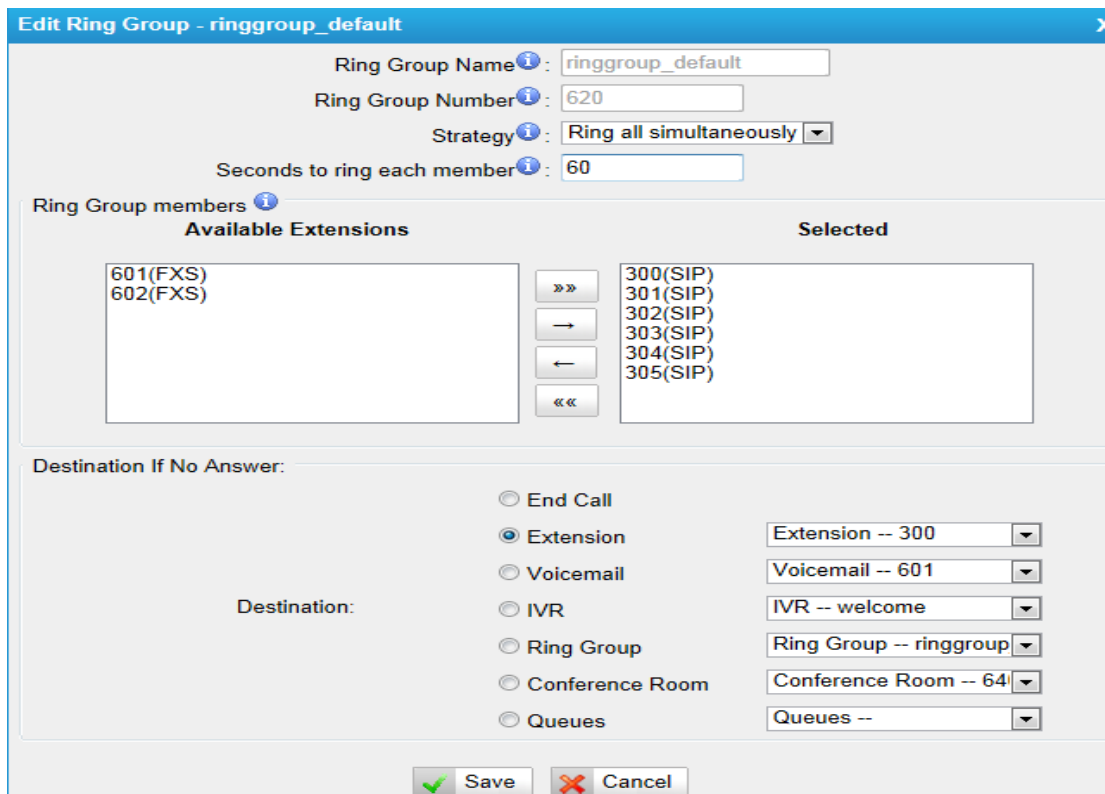


Figure 6-32 Add/Edit Ringgroup

### •Ring Group Name

This option defines a name for this group, i.e. 'Sales'. 'Ring Group Name' is a label to help you identify this group in the group list.

### •Ring Group Number

This option defines the numbered extension that can be dialed to reach this group.

### •Strategy

This option sets the Ringing Strategy for this Group. The options are as follows:

1. RingAll Simultaneously: Ring all available Extensions simultaneously.
2. Ring Sequentially: Ring each extension in the group one at a time.

### •Seconds to ring each member

1. If the strategy is 'Ring All Simultaneously', it means set the number of seconds to ring this group before routing the call according to the 'Destination if No Answer' settings.
2. If the strategy is 'Ring Sequentially', it means set the number of seconds to ring a single extension before moving onto the next one.

### • Ring Group Members

An extension can be made a member of this ring group by moving it into the 'Selected' box.

### •Destination If No Answer

When all members on this group fail to answer the call, system will handle the call according to the selected destination.

## 6.4.3 Queues

Call Queues give users (i.e. call centers) an efficient means to have their calls answered in the order they were received to deliver top tier customer service.



Figure 6-33 Queue Page

Call queues allow calls to be sequenced to one or more agents.

### Notes:

1. Dial 'Queue number + \*' to log in or 'Queue number + \*\*' to log out the queue. For example, if the queue number is '680', then agent can dial '680\*' to log in or '680\*\*' to log out.
2. Follow me feature in extension page will not take effect when it's ringing as an agent of queue

Add Queue
X

Queue Name: 680  
Queue Number: 680  
Queue Password:  
Queue Agent Timeout: 30  
Queue Max Wait Time: 1800  
Queue Ring Strategy: ringall

Agents

Available Agents

Selected

300(SIP)  
301(SIP)  
302(SIP)  
303(SIP)  
304(SIP)  
305(SIP)  
601(FXS)  
602(FXS)

»»  
→  
←  
««

Caller Position Announcements

Announce Position: Yes  
Announce Hold Time: Yes  
Frequency: 30 seconds

Periodic Announcements

Prompt: [Custom Prompts](#)  
Frequency: 30 seconds

Events

Key: ---  
Action: End Call  
Destination:

Failover-Destination

Action: End Call  
Destination:

Others

Music On Hold: calmriver [Music on Hold Prompts](#)  
Leave When Empty: Yes  
Join Empty: No  
Agent Announcement:  
Join Announcement:  
Retry: 30  
Wrap-up Time: 30

Save Cancel

Figure 6-34 Add/Edit Queue

### •Queue Name

A name for the Queue.

### •Queue Number

Use this number to dial into the queue, or transfer callers to this number to put them into the queue.

**•Queue Password**

You can require agents to enter a password before they can login to this queue. This field allowed to be blank empty, it means the agents can login to this queue without password.

**•Queue Agent Timeout**

The number of seconds an agent's phone can ring before we consider it a timeout.

**•Queue Max Wait Time**

The maximum number of seconds a caller can wait in a queue before being pulled out. (0 for unlimited).

**•Queue Ring Strategy**

This option sets the Ringing Strategy for this Queue. The options are

RingAll: Ring All available Agents simultaneously until one answers.

LeastRecent: Ring the Agent which was least recently called.

FewestCalls: Ring the Agent with the fewest completed calls.

Random: Ring a Random Agent.

RRmemory: Round Robin with Memory, Remembers where it left off in the last ring pass.

**1) Agents**

This selection shows all users. Selecting a user here makes them an agent of the current queue.

**2) Caller Position Announcements****•Announce Position**

Announce position of caller in the queue

**•Announce Hold Time**

Enabling this option causes MyPBX to announce the hold time to the caller periodically based on the frequency timer. Either yes or no; hold time will not be announced if <1 minute.

**•Frequency**

How often to announce queue position and estimated hold time.

**Note:** '0 seconds' means disable the announcement

**3) Periodic Announcements****•Prompt**

Select a prompt file to play periodically.

**•Frequency**

How often to announce a prompt to the caller.

4) Events

If a caller presses the key while waiting in the queue, this setting selects which action should process the key press.

5) Failover-Destination

Define the failover action. A failover occurs after the user reach the Queue max wait time.

6) Others

**•Music On Hold**

Select the 'Music on Hold' Class for this Queue.

**•Leave When Empty**

This option controls whether callers already on hold are forced out of a queue that has no agents. There are two options.

Yes: Callers are forced out of a queue when no agents are logged in.

No: Callers will remain in a queue with no agents.

**•Join Empty**

This option controls whether callers can join a call queue that has no agents.

There are two options,

Yes: Callers can join a call queue with no agents or only unavailable agents

No: Callers cannot join a queue with no agents

The default option is No.

**•Agent Announcement**

Announcement played to the Agent prior to bridging in the caller.

**•Join Announcement**

Announcement played to callers once prior to joining the queue.

**•Retry**

The number of seconds we wait before trying all the phones again.

**•Wrap-up time**

How many seconds after the completion of a call an Agent will have before the Queue can ring them with a new call. The default is 30.



## 6.4.4 Conferences



Conference Room	Admin	PIN #		
640	---	No Password		
641	---	No Password		
642	---	No Password		
643	---	No Password		
644	---	No Password		

Figure 6-35 Conference Room List

Conference Calls increase employee efficiency and productivity, and provide a more cost-effective way to hold meetings. Conference agents can dial \* to access to the settings options and the admin can kick the last user out and can lock the conference room.

### •Extension

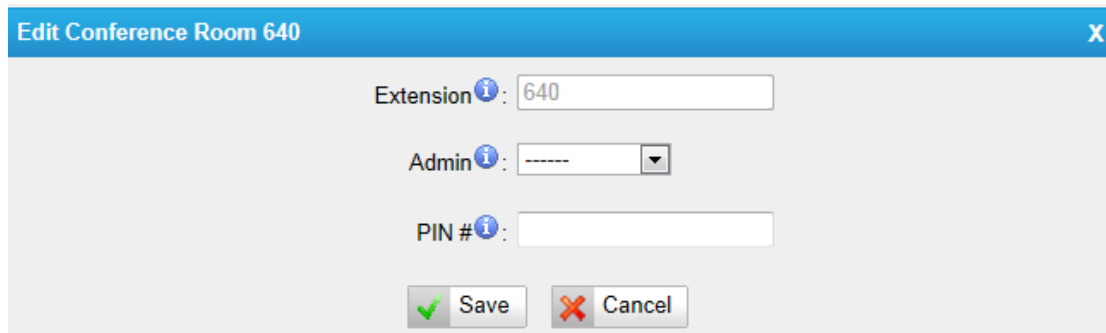
This is the number dialed to reach this Conference Room.

### •Admin

Admin can kick a user out and can lock the conference room.

### •Pin #

Set a PIN # that must be entered in order to access this conference room (i.e. 1234).



**Edit Conference Room 640** X

Extension :

Admin :

PIN # :


Save Cancel

Figure 6-36 Add/Edit Conference Room

## 6.4.5 Inbound Routes

Inbound routing processes incoming call traffic to destination extensions during

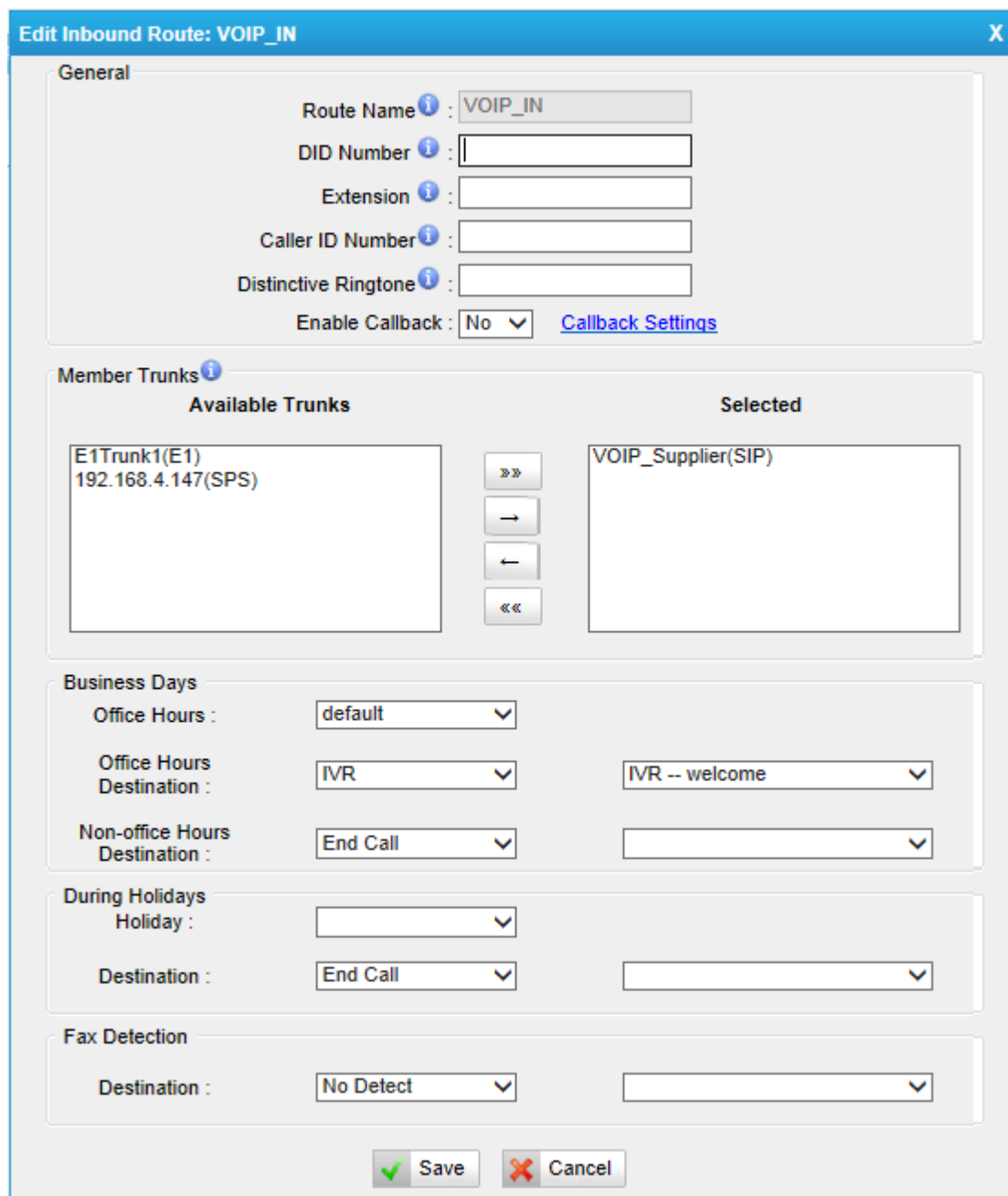
office hours or outside office hours



Route Name	DID Number	Caller ID Number
gstrin		

Figure 6-37 Inbound Route List

There is a default inbound route for all the trunks and set IVR as the destination, you can edit it or create a new one for your demands. When an incoming call arrives, the system will first check 'fax detection', then 'Holidays', at last 'Business Days'.



**Edit Inbound Route: VOIP\_IN**

**General**

Route Name: VOIP\_IN

DID Number:

Extension:

Caller ID Number:

Distinctive Ringtone:

Enable Callback: No [Callback Settings](#)

**Member Trunks**

**Available Trunks**

E1Trunk1(E1)  
192.168.4.147(SPS)

**Selected**

VOIP\_Supplier(SIP)

**Business Days**

Office Hours: default

Office Hours Destination: IVR IVR -- welcome

Non-office Hours Destination: End Call

**During Holidays**

Holiday:

Destination: End Call

**Fax Detection**

Destination: No Detect

figure 6-38 Add/Edit Inbound Route

## 1) General

### •Route Name

A name for this inbound route. Ex: 'VOIP\_IN' etc.

### •DID Number

Define the expected DID Number if this trunk passes DID on incoming calls. Leave this field blank to match calls with any or no DID info. You can also use pattern matching to match a range of numbers. The following patterns may be used:

**X**: Any Digit from 0-9

**Z**: Any Digit from 1-9

**N**: Any Digit from 2-9

**[12345-9]**: Any digit in the brackets (in this example, 1,2,3,4,5,6,7,8,9)

The '.'Character will match any remaining digits. For example, 9011. will match any phone number that starts with 9011, excluding 9011 itself.

The '!' will match none remaining digits, and causes the matching process to complete as soon as it can be determined that no other matches are possible.

Example 1: **NXXXXXX** will match any 7 digits phone number.

Example 2: **1NXXNXXXXXX** will match a phone number starting with a 1, followed by a 3-digit area code, and then 6 digit number.

For more information, please refer to [Appendix F How to Use DID.](#)

### •Extension

Define the extension for DID number. This field is only valid when you use BRI, SIP, SPS or SPX trunk for this inbound router. You can only input number and '-' in this field, and the format can be xxx or xxx-xxx. The count of the number must be only one or equal the count of the DID number.

### •Caller ID Number

Define the Caller ID Number to be matched on incoming calls. Leave this field blank to match any or no DID info.

You can also use a pattern match (e.g. 2[345]X) to match a range of numbers. The following patterns may be used:

**X**: Any Digit from 0-9

**Z**: Any Digit from 1-9

**N**: Any Digit from 2-9

**[12345-9]**: Any digit in the brackets (in this example, 1,2,3,4,5,6,7,8,9)

The '.'Character will match any remaining digits. For example, 9011. will match any phone number that starts with 9011, excluding 9011 itself.

The '!' will match none remaining digits, and causes the matching process to complete as soon as it can be determined that no other matches are possible.

Example 1: **NXXXXXX** will match any 7 digits phone number.

Example 2: **1NXXNXXXXXX** will match a phone number starting with a 1, followed by a 3-digit area code, and then 6 digit number.

#### •Distinctive Ringtone

MyPBX support mapping to custom ring tone files. For example, if you configure the distinctive ringing for custom ring tone to '**Family**', the ring tone will be played if the phone receives the incoming call. This feature works with all IP phones which support distinctive ringtone.

How do I configure distinctive ring tones? Please refer to [APPENDIXE](#).

**Note:** If you want to use feature Distinctive ringtone, please confirm you're your IP phone support this feature firstly. Currently distinctive ringtone can be compatible with Yealink and Snom phone.

#### •Enable Callback

You can enable the callback function of this inbound route. If you want to configure the callback function, please refer to [chapter 6.7.4](#)

#### 2) Member Trunks

This area allows you to select which trunks will be member trunks for this route. To make a trunk a member of this route, please move it to the 'Selected' box.

#### 3) Business Days

Define where the calls will be routed during Business Days.

#### •Office Hours

Select one defined business days office hours.

#### •Office Hours Destination

Configure where to route the incoming calls during office hours.

#### •End Calls

Route the incoming calls to end calls, System will auto hang-up the call.

#### •Extension

Route the incoming calls to a specific extension.

#### •Voicemail

Route the incoming calls to extension's voicemail.

#### •IVR

Route the incoming calls to a specific IVR.

**·Ring Group**

Route the incoming calls to a specific Ring Group.

**·Conference Room**

Route the incoming calls to a specific Conference Room.

**·DISA**

Route the incoming calls to a specific DISA.

**·Queues**

Route the incoming calls to a specific Queue.

**·Faxes**

Route the incoming faxes to a specific extension's mail address.

Note: This function only supports T.38 faxes.

**·Outbound Routes**

Route the incoming calls to a specific outbound route.

This function is mainly used for the connection of two branches.

For example: Company A locates headquarters in the USA with a branch B in China. A and B both have MyPBX phone systems.

Now if staff of A would like to make a call to a telephone or mobile phone in China from the extension of A but via the FXS line of B, that can be done by this configuration.

**·Non-office Hours Destination**

Configure where to route the incoming calls during non-office hours.

**4) During Holidays**

Define where the calls will be routed during Holidays.

**·Holiday**

Select the defined Holiday to use. When a time is defined in both Business Days and Holidays, it will be treated as Holidays.

**·Destination**

Configure where to route the incoming calls during holidays.

**5) Fax Detection**

Configure if detecting faxes in this inbound route.

**Note:** Please choose IVR as the destination above before configure fax detection (recommend)

**•Destination**

Configure where the faxes will be routed when faxes are detected.

**•No detect**

Do not detect faxes.

**•Custom Email**

Customize an E-mail address to receive the faxes. You should first configure the 'Voicemail Settings->SMTP Settings for Voicemail' correctly before you use this option.

**•Faxes**

Send faxes to an extension. If choosing a FXS extension here, the fax will be sent to the FXS port selected, you should connect a fax machine to this FXS port. If Choosing a VoIP extension, the fax will be sent to the extension's voicemail as an attachment.

**Note:** If you receive faxes with custom Email address, the 'SMTP settings' of 'Voicemail Settings' should be configured successfully in advance. If you receive faxes with E-mail address configured in VOIP extension voicemail, you should first make sure the tested email to your email address works fine.

## 6.5 Audio Settings

It's allowed to customize the prompts in MyPBX, including the Audio In and change the system prompts to your local country.

### 6.5.1 Custom Prompts

We can record or upload the prompts in this page, you can also play it directly to confirm if it's a valid one, you can also download it and save it as an backup



Figure 6-39 Custom Prompts List

#### 1. Record new Prompt

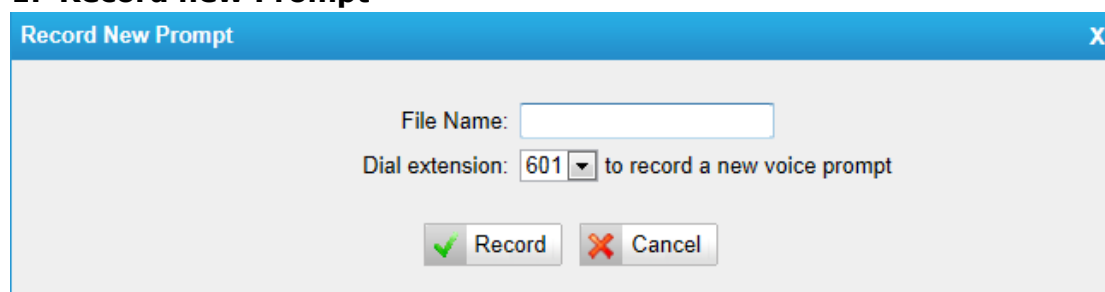


Figure 6-40 Record a New Prompt

The administrator can use this screen to record custom prompts by doing the following:

- 1) Click "Record New Custom Prompt"
- 2) Input the desired file name on the popup window and choose an extension to call for recording (such as 500).
- 3) Click "Record". The selected extension will ring and you can pick up the phone to start recording.

#### 2. Upload Prompt

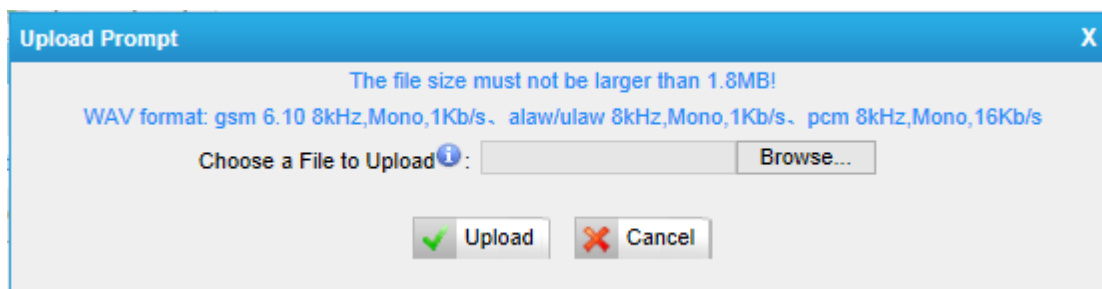


Figure 6-41 Upload a Prompt

The administrator can also upload prompts by doing the following:

- 1) Click "Upload Prompt".
- 2) Click "Browse" to choose the desired prompt.
- 3) Click "Upload" to upload the selected prompt.

**Note:** The file size must not be larger than 1.8 MB, and the file must be WAV format:

GSM 6.10 8 kHz, Mono, 1 Kb/s;

Alaw/Ulaw 8 kHz, Mono, 1 Kb/s;

PCM 8 kHz, Mono, 16 Kb/s.

## 6.5.2 Music on Hold Prompts

In this page, we can upload the music on hold prompts or adjust the volume from Audio In interface (available in MyPBXU300)



Figure 6-42 Music on Hold List

The administrator can upload on hold music as follows:

- 1) Click "Upload Music on Prompt".
- 2) Click "Browse" to choose the desired audio file.
- 3) Click "Upload" to upload the selected file.



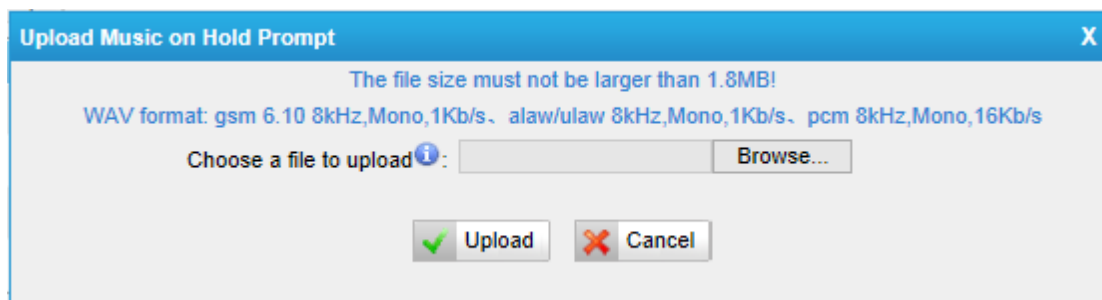


Figure 6-43 Upload Music on Hold

**Note:** The file size must not be larger than 1.8 MB, and the file must be WAV format:

GSM 6.10 8 kHz, Mono, 1 Kb/s;

Alaw/Ulaw 8 kHz, Mono, 1 Kb/s;

PCM 8 kHz, Mono, 16 Kb/s.

### 6.5.3 System Prompts Settings

MyPBX have prompts of many languages. You can download the appropriate language you need. MyPBX can support American English, Australian English, Chinese, Dutch, French, Canadian French, German, Greek, Hungarian, Italian, Polish, Portuguese, Brazilian Portuguese, Russian, Spanish, Mexican Spanish, Turkish, Thai, and Korean currently.

**Note:**

1. Auto-detection is highly recommended. But if you prefer to download via HTTP or TFTP server, please contact the local dealer for the prompts
2. When update successfully, just click "apply the changes" on web then it will take effect, there is no need to reboot.

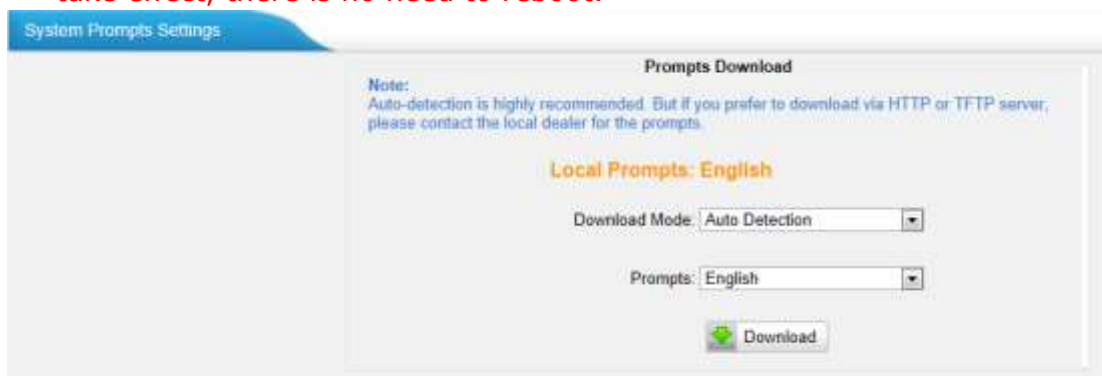


Figure 6-44 System Prompts Settings Page

## 6.6 Basic Settings

There are some basic settings we need to configure MyPBXU300, like the general preferences, business hours, feature codes, voicemail settings

### 6.6.1 General Preferences

In this page, there are some general settings of MyPBX




Figure 6-45 General Settings

#### 1) General

##### •Ring Timeout

Number of seconds to ring a device before handling the call as per the extension's Follow Me settings. Default value is 30s.

##### •MAX call duration

The absolute maximum amount of time permitted for a call. A setting of 0 disables the timeout. Default value is 6000s.

##### •Maximum concurrent calls

Maximum concurrent calls limits. Default value 0 means no limit

##### •Music on hold

Used to set hold music for the system.

**Note:** if you need use the "live music" from "Audio In" interface, please choose "external music" here.

#### •Tone Region

Please select your country or nearest neighboring country to enable the default dial tone, busy tone, and ring tone for your region.

**Note:** please reboot the system to take effect.

#### •HTTP bind port/Web Access Port

Port to use for HTTP sessions; Default: 80

**Note:** please reboot the system to take effect.

#### •FXO Mode

FXO port's operation mode.

#### •Virtual Ring Back Tone

It's only for GSM/UMTS /UMTS trunk. Once enabled, when the caller call out with GSM/UMTS trunks, the caller will only hear the virtual ring back tone generated by the system before callee answers the call.

#### •Distinctive Caller ID

When incoming calls are routed from ring group/queue/IVR, the caller ID displays with the name of ring group/queue/IVR, for example 5503302(ringgroup\_default).

**Note:** To display IVR's name, please press the key instead of the extension number directly.

#### •Attended Transfer CallerID

When transferring an incoming call using the attended transfer feature code or the transfer key of IP phone, the Caller ID of transferee or transferer displayed on the screen of the callee. The default display is the Caller ID of the initiator. For example, if extension 500 makes a call to extension 501. After 501 picks up the call, 501 makes an attended transfer to extension 502. If selecting "Transferer", 502 will display the Caller ID as 500; if selecting "Transferee", 502 will display the Caller ID as 501.

#### •Follow Me Prompt

When set "Follow me" to "Transfer to number" on the extension page (e.g. when 500 is busy, transfer to 501), while 500 is busy, the call will be transferred to 501. If "Enable Follow Me Prompt" choosing yes, there will be prompt before transferring the call. Otherwise, the call will be transferred directly without any prompt. Default: Yes.

**•Music on Hold for Follow Me Prompt**

Configure whether to play a prompt “please hold while I try to locate the person you are calling” when transfer a call by follow me settings.

**•Invalid Phone Number Prompt**

Configure the prompt when the dialed phone number is invalid.

**•Busy Line Prompt**

Configure the prompt when the dialed phone number is busy.

**•Dial Failure Prompt**

Configure the prompt when dial failed due to conjunction no-available channel.

**2) Extension Preferences****•User Extensions**

The default value is 500 to 616

**•Ring Group Extensions**

The default value is 620 to 629

**•Paging Group Extensions**

The default value is 630 to 639

**•Conference Extensions**

The default value is 640 to 659

**•IVR Extensions**

The default value is 660 to 679

**•Queue Extensions**

The default value is 680 to 689

## **6.6.2 Business Hours**

Business hours including “holiday configuration” is used to control the incoming calls, we can configure it in this page.

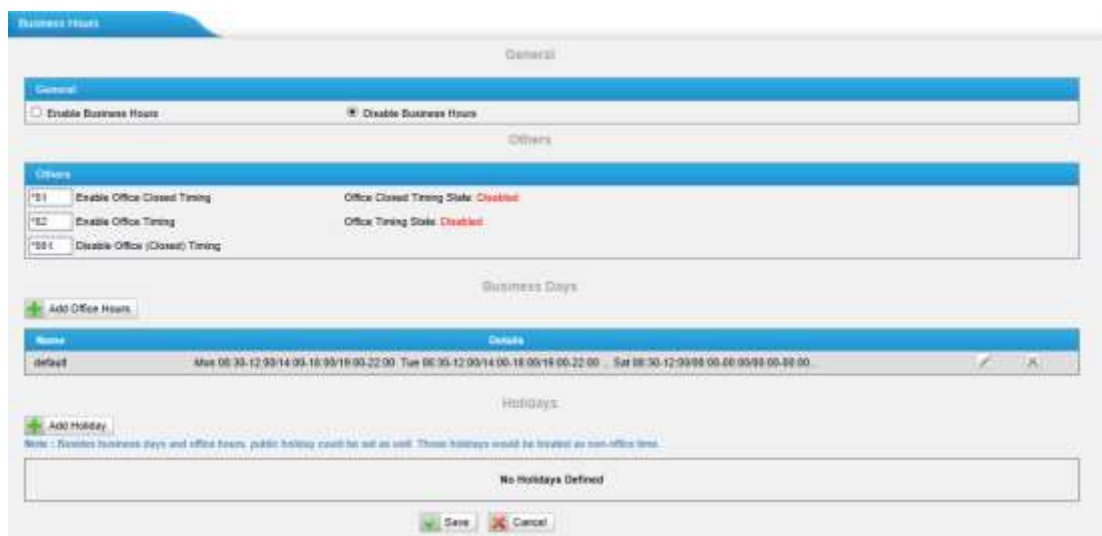


Figure 6-46 Business Hours Settings

### 1) General

#### •Enable Business Hours

#### •disable Business Hours

### 2) Others

#### •Enable Office Closed Timing

By dialing \*81 (\*81 is default) on an extension will force the office time closed for the device whatever the general setting is.

#### •Enable Office Timing

By dialing \*82 (\*82 is default) on an extension will force the office time to take effect for the device whatever the general setting is.

#### •Disable Office closed timing

By dialing \*081 (\*081 is default) on an extension will disable the Office Closed Timing.

### 3) Add office hours

You can setup the business hours here.

### 4) Add Holiday

You can setup the holidays here.

If a time period is configured as both Holidays and office hours, it will be treated as Holidays.

## 6.6.3 Feature Codes

There are many feature codes available in MyPBX, which allow users to dial from extension side to realize the exact feature.



Feature	Feature Code
<input checked="" type="checkbox"/> One Touch Record	*1
<input checked="" type="checkbox"/> Check Extension Voicemail	*2
<input checked="" type="checkbox"/> Voicemail for Extension	#
<input checked="" type="checkbox"/> Voicemail Main Menu	*02
<input checked="" type="checkbox"/> Attended Transfer	*3
<input checked="" type="checkbox"/> Attended Transfer Timeout	15
<input checked="" type="checkbox"/> Blind Transfer	*03
<input checked="" type="checkbox"/> Call Pickup	*4
<input checked="" type="checkbox"/> Extension Pickup	*04
<input checked="" type="checkbox"/> Intercept	*5
<input checked="" type="checkbox"/> Normal Spy	*06
<input checked="" type="checkbox"/> Whisper Spy	*01
<input checked="" type="checkbox"/> Barge Spy	*02
<b>Call Parking Preferences</b>	
Call Parking	*6
Extension range used to park calls	600-699 (Ext: 600-699)
Number of seconds a call can be parked for	60
<b>Call Forwarding Preferences</b>	
<input checked="" type="checkbox"/> Reset to Defaults	*70
<input checked="" type="checkbox"/> Enable Forward All Calls	*71
<input checked="" type="checkbox"/> Disable Forward All Calls	*071
<input checked="" type="checkbox"/> Enable Forward When Busy	*72
<input checked="" type="checkbox"/> Disable Forward When Busy	*072
<input checked="" type="checkbox"/> Enable Forward No Answer	*73
<input checked="" type="checkbox"/> Disable Forward No Answer	*073
<input checked="" type="checkbox"/> Forward to Number	*74
<input checked="" type="checkbox"/> Forward to Voicemail	*074
<input checked="" type="checkbox"/> Enable Do Not Disturb	*75
<input checked="" type="checkbox"/> Disable Do Not Disturb	*075

Figure 6-47 Feature Code Settings Page

### 1) General

#### •One Touch Record

A user may initiate or stop call recording by dialing \*1 during a call. (\*1 is default setting).

#### •Extension for Checking Voicemail

Users can check their Voicemail by dialing \*2 on their phone (\*2 is default setting).

#### •Voicemail for Extension

Users can leave a voicemail to other extensions by dialing # on their phone or the incoming call could be forwarded to an extension's voicemail directly. (# is default setting).

For example, extension 500 want to leave a message for extension 501, users can use 500 dial '#501' to enter the voicemail of 501.

#### •Voicemail main menu

Users can go to the main menu by dialing \*02 (\*02 is default setting).

#### •Attended Transfer

Users may transfer an incoming call by dialing \*3 on their phone (\*3 is default

setting).

**•Attended Transfer Timeout**

The time out of transferring a call

**•Blind Transfer**

Users may blind transfer an incoming call by dialing \*03 on their phone (\*03 is default setting).

**•Call Pickup**

Users may pick up an incoming call by dialing \*4 on their phone (\*4 is default setting)

**•Extension Pickup**

Users may pick up a specific extension's incoming call by dialing \*04+extension number on their phone (\*04 is default setting)

**•Intercom**

Define the feature code that is used to dial an extension in intercom mode. For instance, setting this value to \*5 would allow you to initiate an intercom call with extension 501 by dialing \*5501.

**•Normal Spy**

In this mode, you can only listen to the extension being spied, for example you can dial \*90501 to monitor extension 501

**•Whisper Spy**

In this mode you can listen/whisper to the extension being spied, for example, dialing \*91501 to listen to extension 501, you can also talk with 501 too.

**•Barge Spy**

In this mode, you can barge in both extensions involved in the call, for example dialing \*92501 to barge in and talk with extensions on both sides.

**2) Call Park Preferences****•Call Parking**

User may park an incoming call on his own telephone by pressing ``\*6`` (\*6 is default setting)

**•Extension range used to park calls**

User may park an incoming call on a designated extension at first and then pick up the call again on any other extension.

**•Number of seconds a call can be parked before it is recalled.**

Define the time (in seconds) that a call can be parked before it is recalled to the

station that parked it.

### 3) Call Forwarding Preferences

#### •Reset to Defaults

Users may reset all call forwarding defaults by calling \*70 on their phone (\*70 is default setting).

**Note:** When reset to defaults. The call forwarding settings will be configured as follows:

Always forward: Disabled

Busy forward to Voicemail: Enabled

No answer forward to Voicemail: Enabled

Do not disturb: Disabled

#### •Enable Forward All Calls

Users may enable always forward by calling \*71 on their phone (\*71 is default setting)

#### •Disable Forward All Calls

Users may disable always forward by calling \*071 on their phone (\*071 is default setting)

#### •Enable Forward When Busy

Users may enable busy forward by dialing \*72 on their phone (\*72 is default setting)

#### •Disable Forward When Busy

Users may disable busy forward by calling \*072 on their phone (\*072 is default setting)

#### •Enable Forward No Answer

Users may enable no answer forward by calling \*73 on their phone (\*73 is default setting)

#### •Disable Forward No Answer

Users may disable no answer forward by calling \*073 on their phone (\*073 is default setting)

#### •Forward to number

Users may activate call forwarding by dialing this feature code, followed by the extension or phone number to forward all calls to this number.

**Note:** Users may activate Forward to number by dialing \*74 + phone number. e.g.: by dialing \*74501, all calls will be forwarded to extension 501.



#### •Forward to Voicemail

Users may forward the call to Voicemail by calling \*074 on their phone (\*074 is default setting)

#### •Enable Do Not Disturb

Users may enable do not disturb by calling \*75 on their phone (\*75 is default setting)

#### •Disable Do Not Disturb

Users may disable do not disturb by calling \*075 on their phone (\*075 is default setting)

## 6.6.4 Voicemail Settings

In this page, we can configure some settings for voicemail feature, including general voicemail settings and SMTP settings, which is used for "voicemail to email"

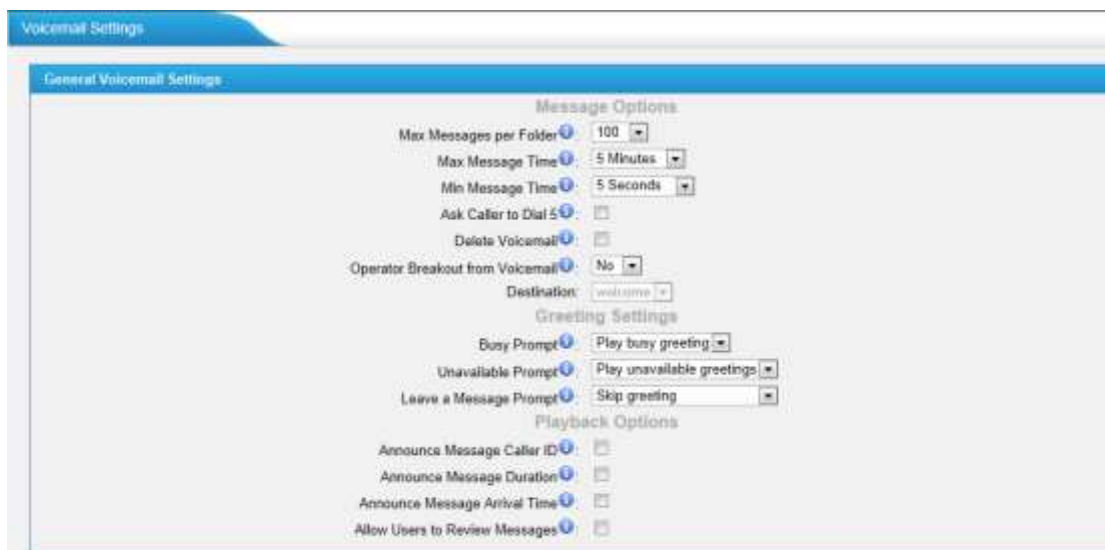


Figure 6-48 General Voicemail Settings

### 1) General Voicemail Settings

#### a) Message Options

##### •Max Messages per Folder

Set the maximum number of messages that can be stored in a single voicemail box.

##### •Max Message Time

Set the maximum length of a single voicemail message.

**•Min Message Time**

Set the minimum length of a single voicemail message. Messages below this threshold will be automatically deleted.

**•Ask Caller to Dial 5**

If this option is set, the caller will be prompted to press 5 before leaving a message.

**•Operator Breakout from Voicemail**

If this option is set, the caller can jump out of the voicemail and go to the destination (IVR) you set by dialing "0".

**b) Greeting Settings****•Busy Prompt**

Greeting played when the extension called is busy.

Skip greeting: Do not play a greeting.

Play busy greeting: play the extension busy greeting.

**•Unavailable Prompt**

Greeting played when the extension called is Unavailable.

Skip greeting: Do not play a greeting.

Play Unavailable greeting: play the extension Unavailable greeting.

**•Leave a Message Prompt**

Greeting played to ask the caller to dial 5 to leave a message.

Skip greeting: Do not play a greeting.

Play busy greeting: play the extension busy greeting.

Play Unavailable greeting: play the extension Unavailable greeting.

**c) Playback Options****•Announce Message Caller ID**

If this option is enabled, the Caller ID of the party that left the message will be played back before the voicemail message begins playing.

**•Announce Message Duration**

If this option is set, the duration of the message in minutes will be played back before the voicemail message begins playing.

**•Announce Message Arrival Time**

If this option is set, the arrival time of the message will be played back before the voicemail message begins playing.

**•Allow Users to Review Messages**

Allow callers to review their recorded message before sending it to voicemail.

## 2) SMTP Settings for Voicemail

**Note:** If you want to send voicemail messages as email attachments, please configure this section.

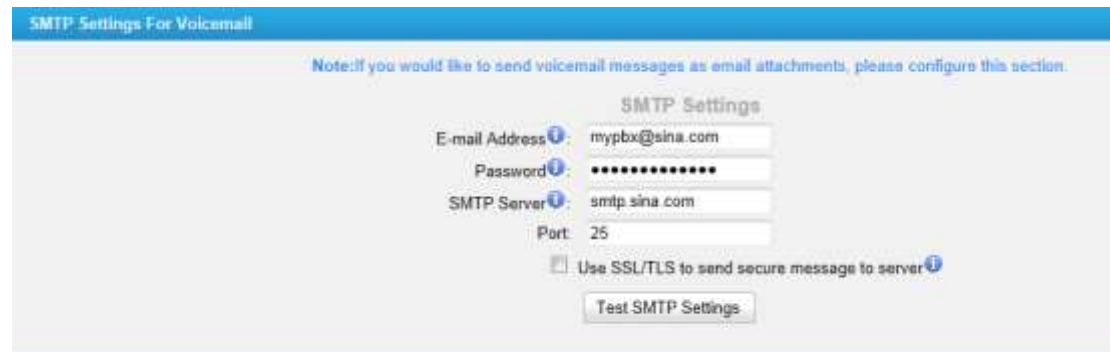


Figure 6-49 SMTP Server Settings

### •E-mail Address

The E-mail Address that MyPBX will use to send voicemail.

### •Password

The password for the email address used above

### •SMTP Server

The IP address or hostname of an SMTP server that the MyPBX will connect to in order to send voicemail messages via email, i.e. mail.yourcompany.com.

### •Port

SMTP Port: the default value is 25.

### •Use SSL/TLS to send secure message to server

If the server of sending email needs to authenticate the sender, you need to select the check box.

**Notes:** Must be selected for Gmail or exchange server.

After filling out the above information, you can click on the "Test Account Settings" button to check whether the setup is OK.

1) If the test is successful, you can use the email safely.

2) If test failed, please check if the above information is correct or network is proper.

## 6.7 Advanced Settings

### 6.7.1 SIP Settings

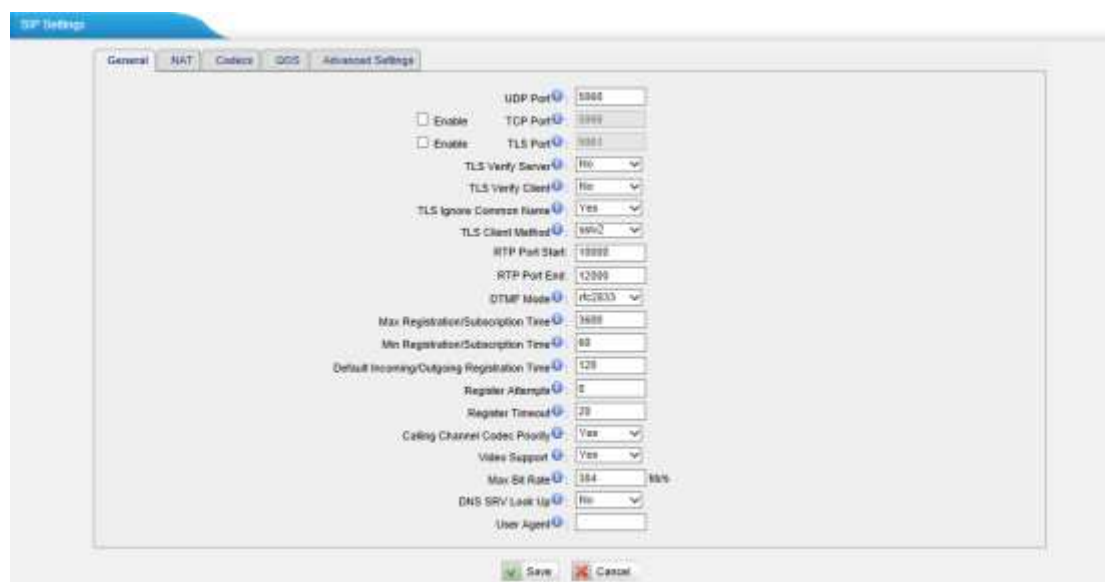


Figure 6-50 SIP Settings-General

#### 1) General

##### •UDP Port

Port use for sip registrations, Default is 5060.

##### •TCP Port

Port use for sip registrations, Default is 5060.

##### •TLS Port

Port use for sip registrations, Default is 5061.

##### •TLS Verify Server

When using MyPBX as a TLS client, whether or not to verify server's certificate. It is "No" by default.

##### •TLS Verify Client

When using MyPBX as a TLS server, whether or not to verify client's certificate. It is "No" by default.

##### •TLS Ignore Common Name

Set this parameter as "No", then common name must be the same with IP or domain name.

##### •TLS Client Method

When using MyPBX as TLS client, specify the protocol for outbound TLS connections. You can select it as tlsv1, sslv2 or sslv3.

**•RTP Port Start**

Beginning of RTP port range

**•RTP Port End**

End of RTP port range

**•DTMF Mode**

Set default mode for sending DTMF. Default setting: rfc2833

**•Max Registration/Subscription Time**

Maximum duration (in seconds) of a SIP registration. Default is 3600 seconds.

**•Min Registration/Subscription Time**

Minimum duration (in seconds) of a SIP registration. Default is 60 seconds.

**•Default Incoming/Outgoing Registration Time**

Default Incoming/Outgoing Registration Time: Default duration (in seconds) of incoming/outgoing registration.

**•Register Attempts**

The number of SIP REGISTER messages to send to a SIP Registrar before giving up. Default is 0 (no limit).

**•Register Timeout**

Number of seconds to wait for a response from a SIP Registrar before timed out. Default is 20 seconds.

**•Calling Channel Codec Priority**

Once enabled, when dialing out via SIP/SPS trunks, the codec of calling channel will be selected in preference. If not, MyPBX will follow the priority in your SIP/SPS trunks.

**•Video Support**

Support for SIP video or no. Default is yes.

**•Max Bit Rate**

Configure the max bit rate for video stream. The default: 384kb/s

**•DNS SRV Look Up**

Please enable this option when your SIP trunk contains more than one IP address.

## •User Agent

To change the user agent parameter of asterisk, the default is "MyPBX", you should change it if needed.

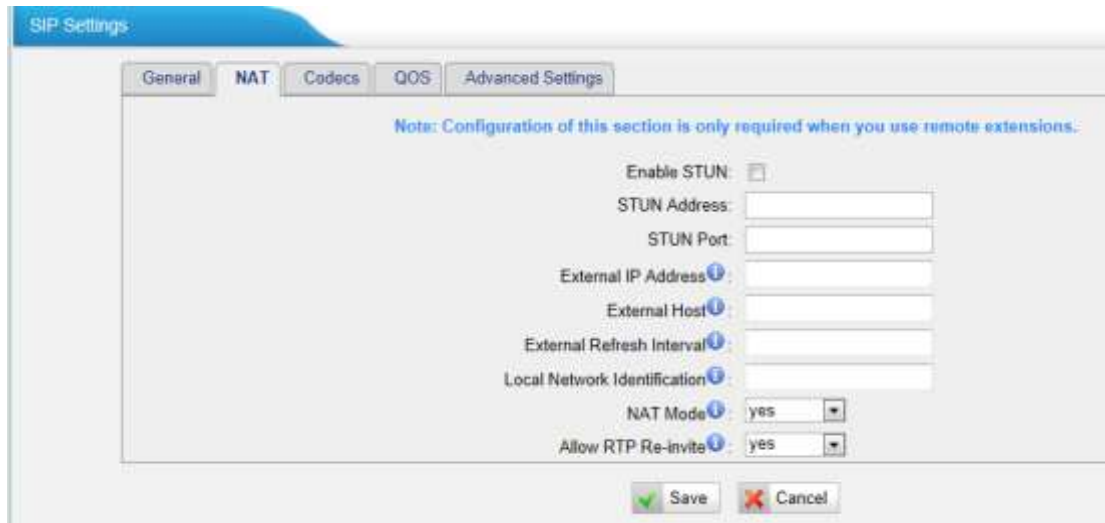


Figure 6-51 SIP Settings-NAT

## 2) NAT

**Note:** Configuration of this section is only required when using remote extensions.

## •Enable STUN

STUN (Simple Traversal of UDP through NATs) is a protocol for assisting devices behind a NAT firewall or router with their packet routing.

## •STUN Address

The STUN server allows clients to find out their public address, the type of NAT they are behind and the internet side port associated by the NAT with a particular local port. This information is used to set up UDP communication between the client and the VOIP provider and so establish a call.

## •External IP Address

The IP address that will be associated with outbound SIP messages if the system is in a NAT environment.

## •External Host

Alternatively you can specify an external host, and the system will perform DNS queries periodically.

This setting is only required when your public IP address is not static. It is recommended that a static public IP address be used with this system. Please contact your ISP for more information.

## •External Refresh Interval

If an external host has been supplied, you may specify how often the system will perform a DNS query on this host. This value is specified in seconds.

### •Local Network Identification

Used to identify the local network using a network number/subnet mask pair when the system is behind a NAT or firewall.

Some examples of this are as follows:

"192.168.0.0/255.255.0.0": All RFC 1918 addresses are local networks;

"10.0.0.0/255.0.0.0": Also RFC1918;

"172.16.0.0/12": Another RFC1918 with CIDR notation;

"169.254.0.0/255.255.0.0": Zero conf local network.

Please refer to RFC1918 for more information.

### •NAT Mode

Global NAT configuration for the system; the options for this setting are as follows:

Yes = Use NAT. Ignore address information in the SIP/SDP headers and reply to the sender's IP address/port.

No = Use NAT mode only according to RFC3581.

Never = Never attempt NAT mode or RFC3581 support.

Route = Use NAT but do not include rport in headers.

### •Allow RTP Reinvite

By default, the system will route media streams from SIP endpoints through itself. Enabling this option causes the system to attempt to negotiate the endpoints to route packets to each other directly, bypassing the system. It is not always possible for the system to negotiate endpoint-to-endpoint media routing.

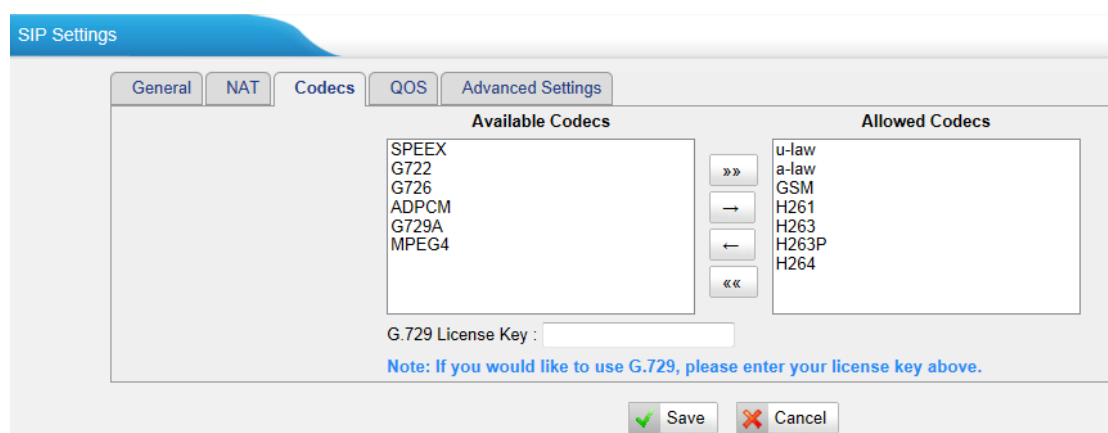


Figure 6-52 SIP Settings-Codecs

## 3) Codecs

A codec is a compression or decompression algorithm that used in the

transmission of voice packets over a network or the Internet.

**u-law:** A PSTN standard codec, used in North America, which provides very good voice quality and consumes 64kbit/s in each direction (receiving and transmitting) of a VoIP call.

**a-law:** A PSTN standard codec, used outside of North America, which provides very good voice quality and consumes 64kbit/s in each direction (receiving and transmitting) of a VoIP call.

**GSM/UMTS:** A wireless standard codec, used worldwide, that provides adequate voice quality and consumes 13.3kbit/s in each direction (receiving and transmitting) of a VoIP call. GSM/UMTS is supported by many VoIP phones.

**SPEEX:** Speex is an Open Source/Free Software patent-free audio compression format designed for speech. The Speex Project aims to lower the barrier of entry for voice applications by providing a free alternative to expensive proprietary speech codecs. Moreover, Speex is well-adapted to Internet applications and provides useful features that are not present in most other codecs.

**G.722:** G.722 is a wideband speech coding algorithms which supports the bit rate of 64, 56 and 48kbps wideband. It's a broadband voice encoding of G series.

**G.726:** A PSTN codec, used worldwide, that provides good voice quality and consumes 32kbit/s in each direction (receiving and transmitting) of a VoIP call. G.726 is supported by some VoIP phones.

**ADPCM, G.729A, H261, H263, H263p, H264, MPEG4.**

**Note:** If you would like to use G.729, please enter your license.

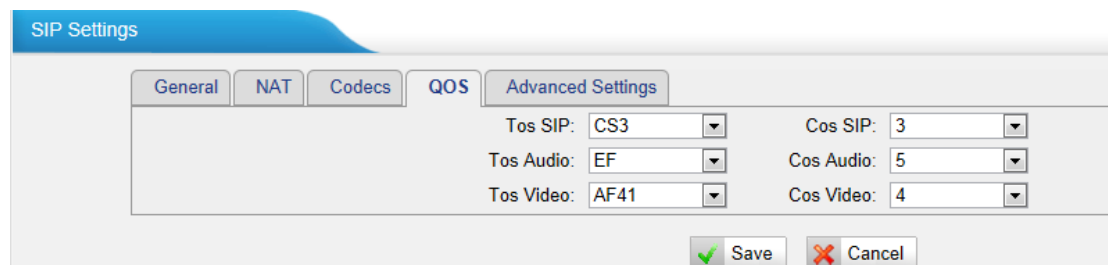


Figure 6-53 SIP Settings-QoS

#### 4) QoS

QoS (Quality of Service) is a major issue in VoIP implementations. The issue is how to guarantee that packet traffic for a voice or other media connection will not be delayed or dropped due interference from other lower priority



traffic. When the network capacity is insufficient, QoS could provide priority to users by setting the value.

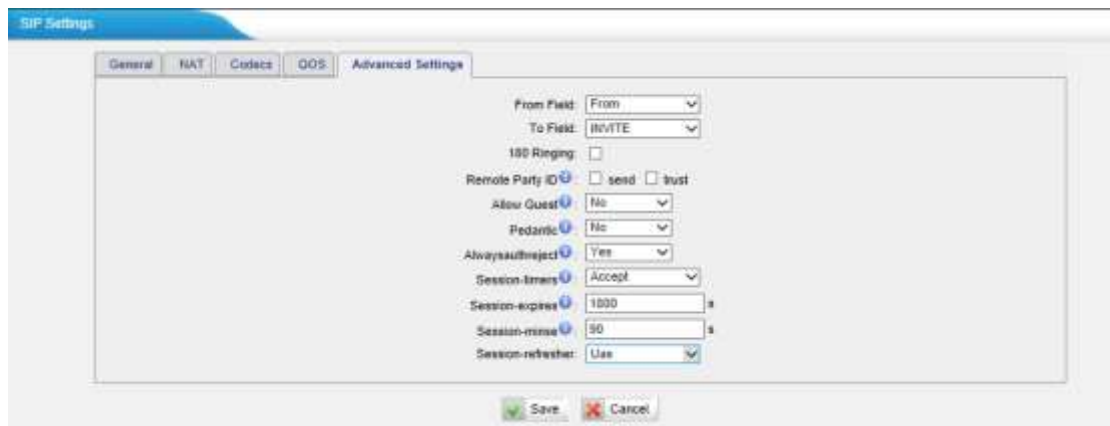


Figure 6-54 SIP Settings-Advanced Settings

## 5) Advanced Settings

### •From Field

Where to get the caller ID in sip packet.

### •To Field

Where to get the DID in sip packet.

### •180 Ringing

It is set when the telecom provider needs. Usually it is not needed.

### •Remote Party ID

Whether send Remote-Party-ID on SIP header. Default no.

### •Allow Guest

Whether allow anonymous registration extension. Default: no.

This option is used to avoid some anonymous calls by hackers. For more details about the system security configuration, please refer to [APPENDIX G MyPBX Security Configuration Guide](#).

### •Pedantic

Enable pedantic parameter. Default: no.

### •Alwaysauthreject

If enabled, when MyPBX rejects "Register" or "Invite" packets, MyPBX always respond the packets using "SIP404 NOT FOUND".

### •Session -timers

Enable session-timer mode, default: yes

### •Session-expires

The max refresh interval

### •Session-minse

The min refresh interval, which mustn't be less than 90s

### •Session-refresher

Choose session-refresher, the default is Uas

## 6.7.2 IAX Settings

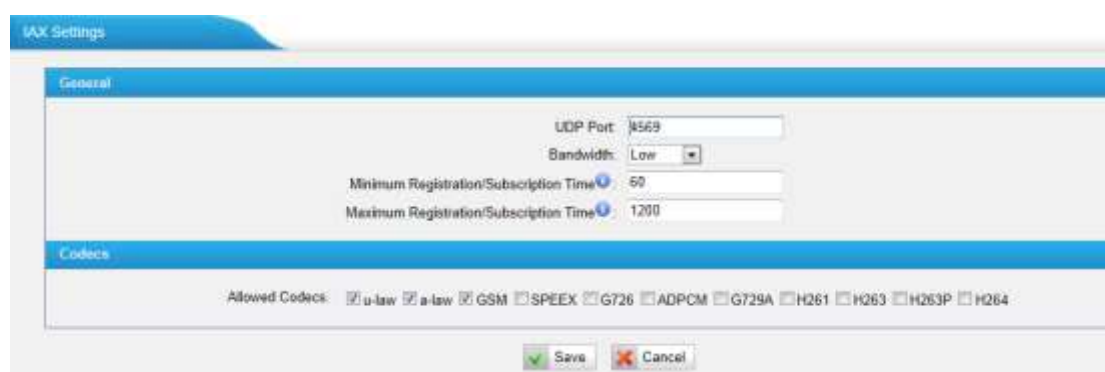


Figure 6-55 IAX Settings

### 1) General

#### •Bind Port

Port use for IAX2 registrations, Default is 4569.

#### •Bandwidth

Low/medium/high with this option you can control which codec to be used.

#### •Min Registration Time

Minimum duration (in seconds) of a IAX2 registration. Default is 60 seconds.

#### •Max Registration Time

Maximum duration (in seconds) of a IAX2 registration. Default is 1200 seconds.

### 2) Codecs

A codec is a compression or decompression algorithm that used in the transmission of voice packets over a network or the Internet.

**u-law:** A PSTN standard codec, used in North America, that provides very good voice quality and consumes 64 kbit/s in each direction (receiving and transmitting) of a VoIP call.

**a-law:** A PSTN standard codec, used outside of North America, that provides very good voice quality and consumes 64kbit/s in each direction (receiving and transmitting) of a VoIP call.

**GSM:** A wireless standard codec, used worldwide, that provides adequate voice quality and consumes 13.3kbit/s in each direction (receiving and transmitting) of a VoIP call. GSM is supported by many VoIP phones.

**SPEEX:** Speex is an Open Source/Free Software patent-free audio compression format designed for speech. The Speex Project aims to lower the barrier of entry for voice applications by providing a free alternative to expensive proprietary speech codecs. Moreover, Speex is well-adapted to Internet applications and provides useful features that are not present in most other codecs.

**G.726:** A PSTN codec, used worldwide, that provides good voice quality and consumes 32kbit/s in each direction (receiving and transmitting) of a VoIP call. G.726 is supported by some VoIP phones.

**ADPCM, G.729A, H261, H263, H263p, H264.**

**Note:** If you would like to use G.729, please enter your license.

### 6.7.3 Blacklist

Blacklist is used to block an incoming/outgoing call. If the number of incoming/outgoing call is registered in the number blacklist, the caller will hear the following prompt: "The number you have dialed is not in service. Please check the number and try again". The system will then disconnect the call.

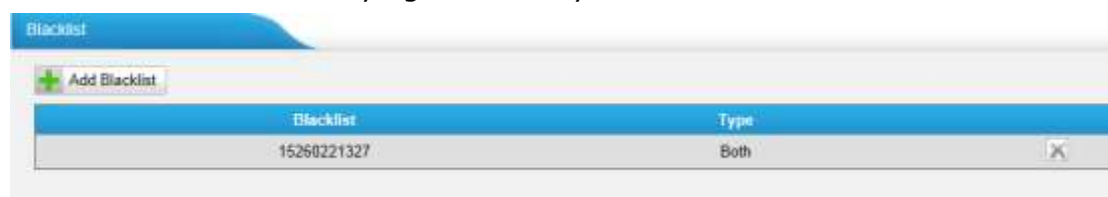


Figure 6-56 Blacklist List

We can add a number with the type: inbound, outbound or both

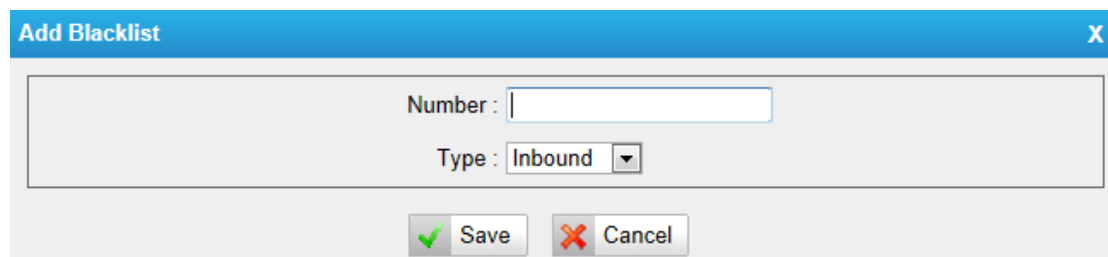


Figure 6-57 Add a Number as Blacklist

**Notes:** Add an extension in blacklist, such as 300:

- 1) If the type is "inbound", then this extension can't dial other extension and external numbers, but it can dial IVR or ringgroup.
- 2) If the type is "outbound", then this extension can't be called.

## 6.7.4 Callback Settings

MyPBX allows caller A to dial an inbound route number, and after hearing the ring, A can hang up the call or wait for MyPBX to cut off the call, then MyPBX will call A with this number. When A picks up the call, A can dial the number he wants to call; MyPBX will call the number with its outbound route.

### Notes:

1. If you'd like to use callback feature, please make sure it's enabled on the inbound route setting panel.
2. No callback rules needed to be set if the trunk supports call back with the caller ID directly.

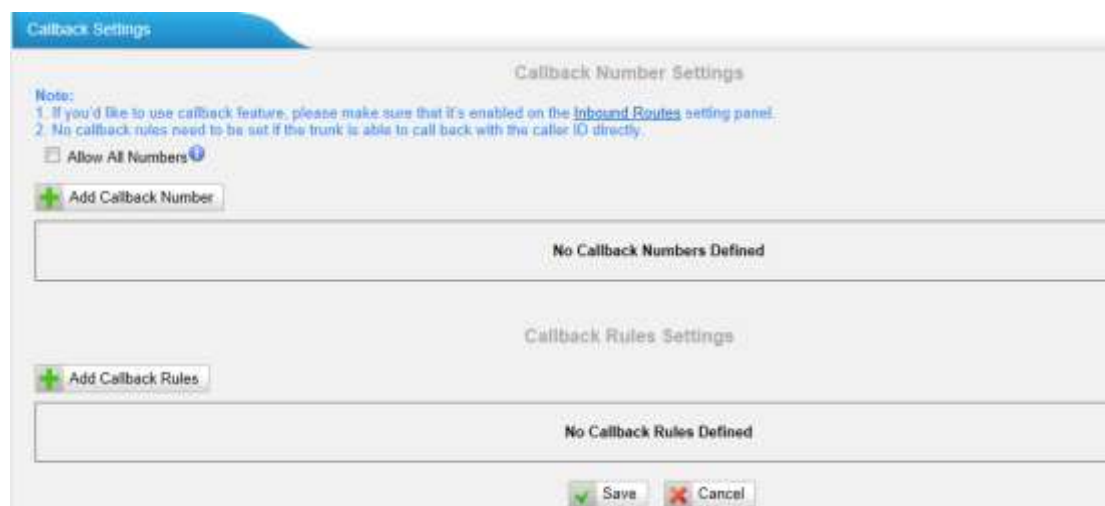


Figure 6-58 Callback Settings Page

### •Allow All Numbers

If you want to apply Callback function to all incoming numbers, please tick Allow All numbers.

Follow the step to use this function.

Step 1: Enable Callback.

Inbound Routes –Choose "Yes" on "Enable Callback" to enable this function

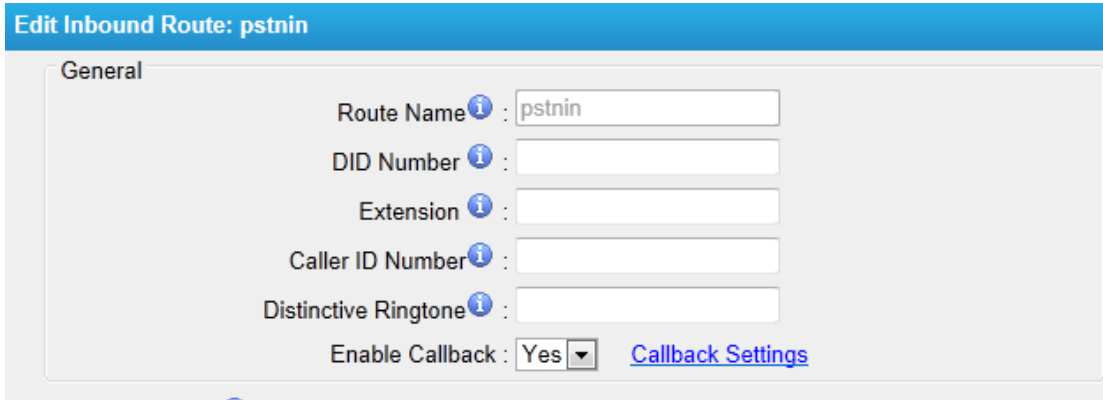


Figure 6-59 Enable Callback in Inbound Route

## Step 2: Create Callback number

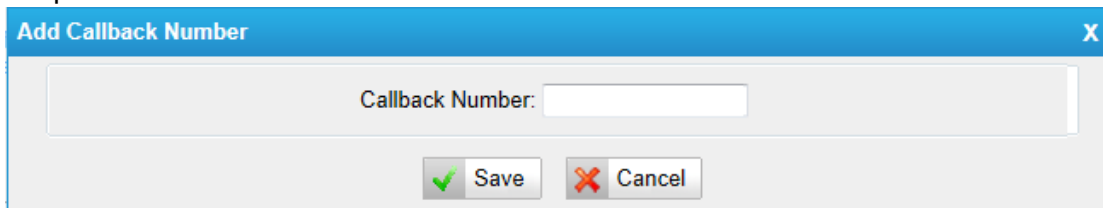


Figure 6-60 Add a New Callback Number

## Step 3: Create Callback Rules

You will need to create callback rules when the system should strip or add digits.

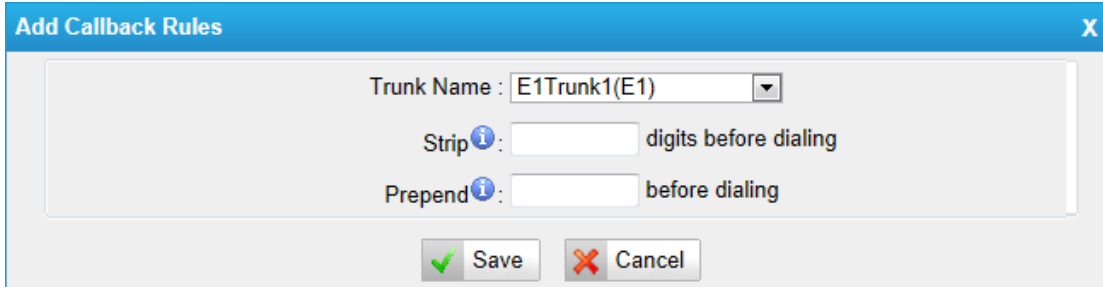


Figure 6-61 Create Callback Rule

### •Trunk Name

Choose the trunk with callback rules

### •Strip digits from front

Define how many digits will be stripped from the call in number before the callback is placed. For example, when you call from number 123456789 into MyPBX, the caller ID is 0123456789, but you can only call 123456789 successfully from MyPBX trunk. You should configure number 0123456789 as the call back number and strip 1 digit before the callback is placed

### •Prepend before dialing

Define digits added before a callback number before the callback is placed. For example, the call in number (Caller ID) is 123456789, MyPBX need to send

9123456789 to its trunk when call to this number. You should configure 123456789 as the call back number and add 9 before the callback is placed. You can add 'w' for analog trunks for some delay too

## 6.7.5 DNIS Settings

DNIS (Dialed Number Identification Service) is a telephone service that identifies for the receiver of a call the number that the caller dialed

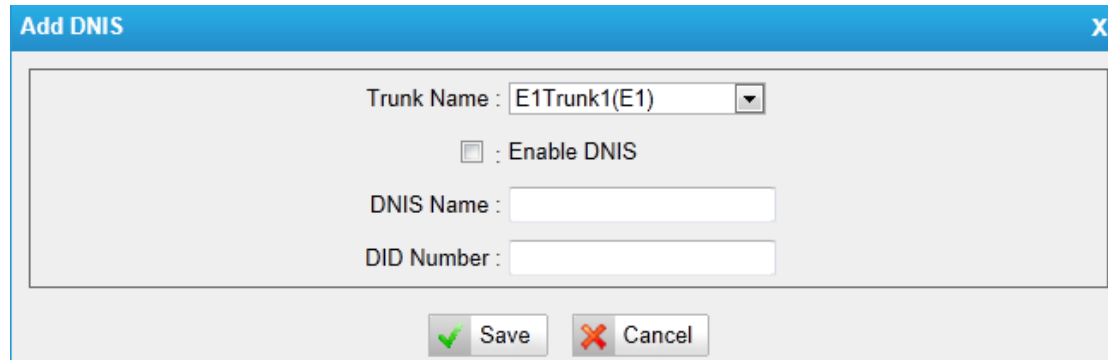


Figure 6-62 Add a New DNIS

**Note:** If DID is not configured here, all the calls via this trunk will show the DNIS instead of the original caller ID

## 6.7.6 DISA

DISA (Direct Inward System Access) allows someone calling in from outside the telephone switch (PBX) to obtain an 'internal' system dial tone and make calls as if they were using one of the extensions attached to the telephone switch. To use DISA, a user calls a DISA number, which invokes the DISA application. The DISA application in turn requires the user to enter a PIN number, followed by the pound sign (#). If the PIN number is correct, the user will hear dial tone on which a call may be placed. Obviously, this type of access has serious security implications, and great care must be taken not to compromise your security.

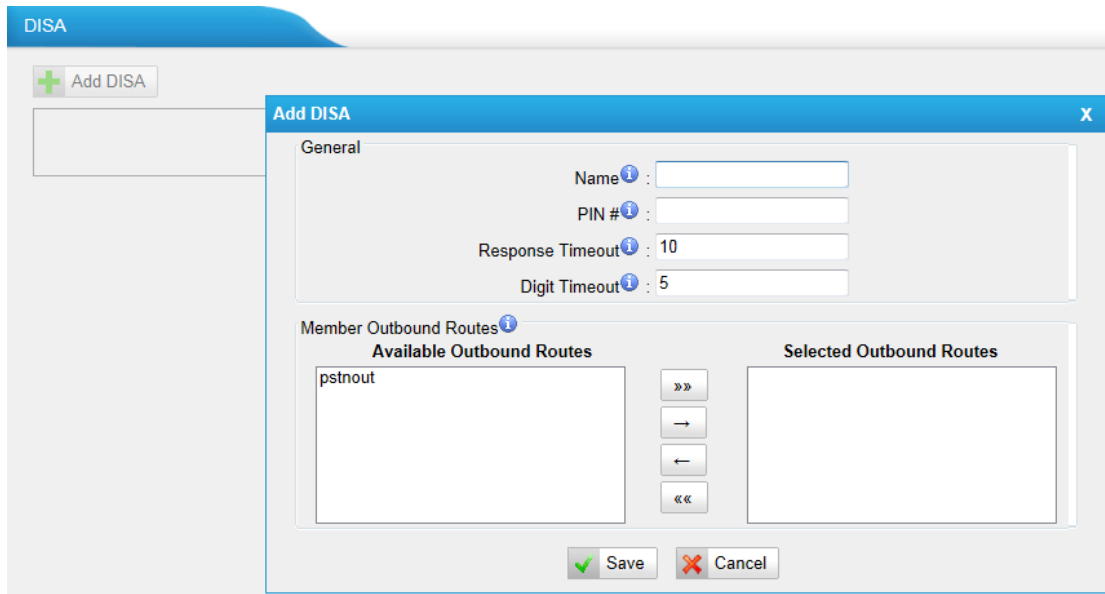


Figure 6-63 Add a New DISA

#### 1) General

##### •DISA Name

Give this DISA application a name to help you identify it.

##### •PIN#

The password for this DISA.

##### •Response Timeout

The maximum amount of time the system will wait before hanging up the call if the user has dialed an incomplete or invalid number. Default is 10 seconds.

##### •Digit Timeout

The maximum amount of time permitted between each digit when the user is dialing an extension number. Default is 5 seconds.

#### 2) Member Outbound Routes

Used to set the outbound routes that can be accessed from this DISA.

## 6.7.5 PIN User Settings

PIN User is used to manage lists of PINs that can be used to access restricted features such as Outbound Routes.



The screenshot shows the 'PIN User Settings' page. At the top, there is a tab labeled 'PIN User Settings'. Below the tab, there is a button labeled '+ Add PIN User'. A message states 'No PIN Users Defined'. Below this, there is an 'Options' section. It contains three fields: 'Access Code' with the value '+89', 'Prompt for Entry' with a dropdown menu showing 'pinuser-entry', and 'Prompt for Failed Entry' with a dropdown menu showing 'pinuser-error'. At the bottom of the options section, there are 'Save' and 'Cancel' buttons.

Figure 6-64 PIN User Settings Page

### 1) Options

#### •Access Code

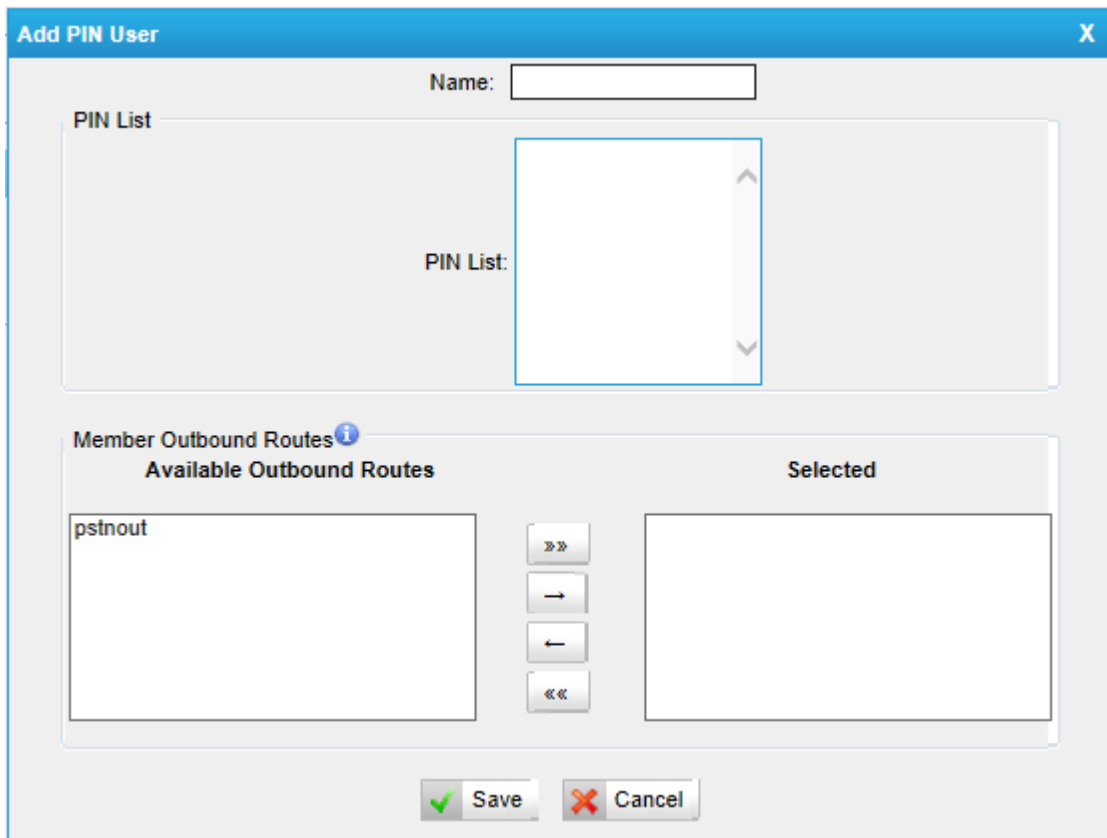
.Dial this code to access PIN.

#### •Prompt for Entry

Prompt caller enter the PIN Number.

#### •Prompt for Entry Failure

Prompt the caller when an invalid PIN is entered.



The screenshot shows the 'Add PIN User' dialog box. It has a title bar with 'Add PIN User' and a close button. Inside, there is a 'Name' field. Below it, there is a 'PIN List' section with a large text area labeled 'PIN List:'. At the bottom, there is a 'Member Outbound Routes' section. It contains two columns: 'Available Outbound Routes' and 'Selected'. The 'Available Outbound Routes' column contains the text 'pstnout'. Between the columns are four buttons: '>>', '→', '←', and '<<'. At the bottom of the dialog, there are 'Save' and 'Cancel' buttons.

Figure 6-65 Add a New PIN User

### 2) PIN User

MyPBX can store a number of PIN Users. PIN Users may be used to keep track of calls in relation to particular activities or clients. They can also be used to



keep track of calls by particular users or sets of users.

- PIN entered are checked against those stored by the system. If an invalid PIN is entered, the PIN is requested again.
- The system administrator can configure certain numbers or types of numbers to require entry of a PIN before you can continue making a call to such a number.
- The system administrator can also configure you to have to enter a PIN before making any external call.

#### •Name

A character-based name for this PIN list, i.e. 'YeastarPIN'

#### •PIN List

Enter a list of one or more PINs, One PIN per line.

#### •Outbound Route

PIN User can use those outbound route to make call out

## 6.7.8 Paging Groups

Paging is used to make an announcement over the speakerphone to a phone or group of phones. Targeted phones will not ring, but instead answer immediately into speakerphone mode. Please note that this section is for configuring paging groups. If you would like to configure Intercom settings, please open the Other Settings -> Feature Codes screen.

This feature is supported by the following SIP phones:

Yealink's T28,T26,T22,T20,T10T,T9CM. Other SIP devices may also work with this feature but are not officially supported.

**Note:**A paging group can have a maximum of 20 members.



Figure 6-66 Paging Group List

There are two types of paging groups in MyPBX U300

#### 1. Internal paging Group

In this mode, if you dial its number, MyPBX will help to pick up those chosen members and you can talk directly without any rings.

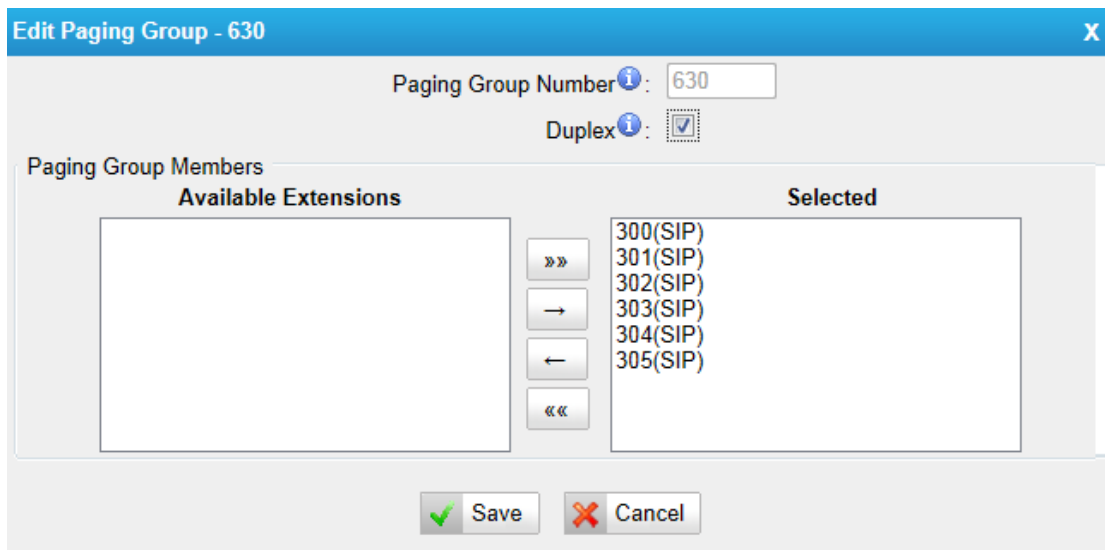


Figure 6-67 Add/Edit Internal Paging Group

### •Paging Group Number

Define the numbered extension that may be dialed to reach this group.

### •Duplex

Paging is typically one way for announcements only. Checking this will make paging duplex, allowing all users in the group to talk and be heard by all.

### 2. External paging group

In this mode, the chosen extensions will have the rights to dial the group number; the voice will be broadcasted via the 'Audio Out' interface of U300

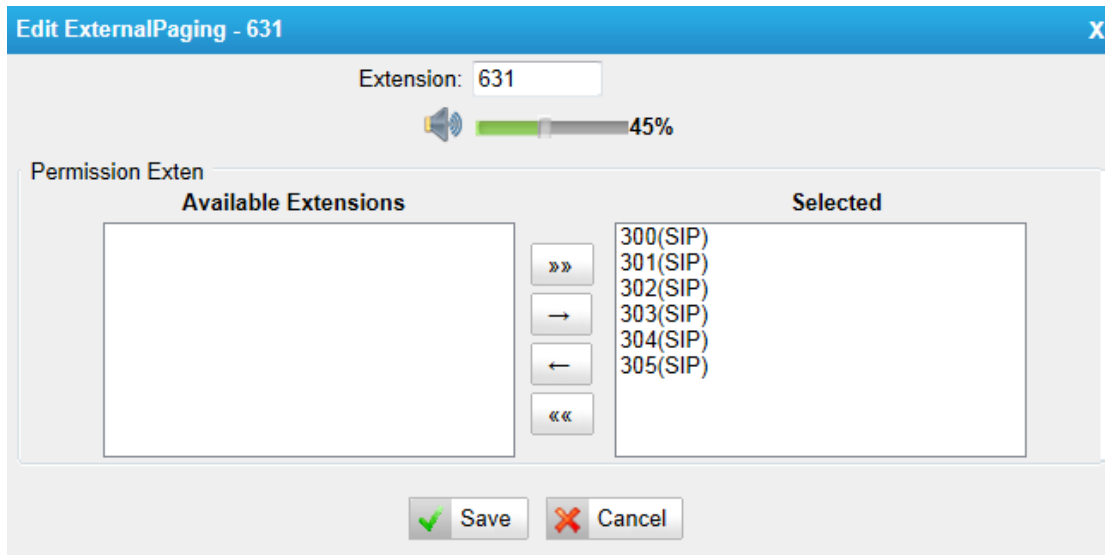


Figure 6-68 External Paging

## 6.7.9 Certificates

MyPBX can support TLS extension. Before you register TLS extension on IP phone, you should upload certificates first.



Figure 6-69 Upload Certificate

### Trusted Certificate

This certificate is a CA certificate. When selecting "TLS Verify Client" as "Yes", you should upload a CA. The relevant IP phone should also have the according server certificate.

### PBX Certificate

This certificate is server certificate. No matter selecting "TLS Verify Client" as "Yes" or "NO", you should upload this certificate to MyPBX. If IP phone enables "TLS Verify server", you should also upload the according CA certificate on IP phone.

## 7 Reports

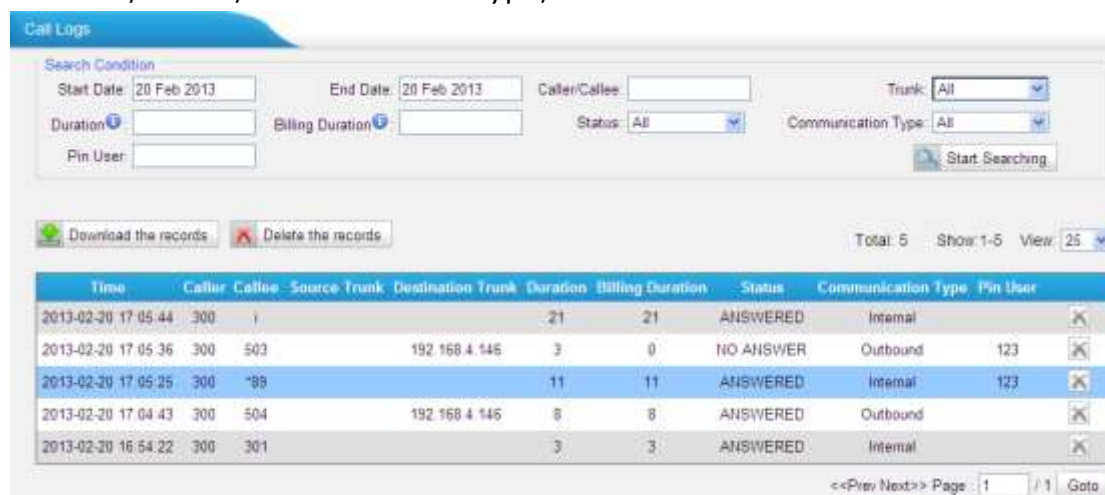


Click  to access

We can check the call detailed logs for counting and system log for debugging

### 7.1 Call Logs

The call Log captures all call details, including call time, caller number, callee number, call type, call duration, etc. An administrator can search and filter call data by filter the call logs by call date, caller/callee, trunk, duration, billing duration, status, communication type, Pin User.



The screenshot shows the 'Call Logs' interface. At the top, there are search filters: Start Date (20 Feb 2013), End Date (20 Feb 2013), Caller/Callee, Trunk (All), Duration, Billing Duration, Status (All), and Communication Type (All). Below the filters are buttons for 'Download the records' and 'Delete the records'. The main area displays a table of call records with columns: Time, Caller, Callee, Source Trunk, Destination Trunk, Duration, Billing Duration, Status, Communication Type, and Pin User. The table shows 5 records, with the third record highlighted. At the bottom, there are navigation controls: '<< Prev Next >>' and 'Page 1 / 1 Goto'.

Time	Caller	Callee	Source Trunk	Destination Trunk	Duration	Billing Duration	Status	Communication Type	Pin User
2013-02-20 17:05:44	300	1			21	21	ANSWERED	Internal	
2013-02-20 17:05:36	300	503		192 168 4 146	3	0	NO ANSWER	Outbound	123
2013-02-20 17:05:25	300	*89			11	11	ANSWERED	Internal	123
2013-02-20 17:04:43	300	504		192 168 4 146	8	8	ANSWERED	Outbound	
2013-02-20 16:54:22	300	301			3	3	ANSWERED	Internal	

Figure 7-1 Call Log List

## 7.2 System Logs



Figure 7-2 System Logs

You can download and delete the system logs of MyPBX.

### Options

#### •Enable Hardware Log

Save the information of hardware; (up to 4 log files)

#### •Enable Normal Log

Save the prompt information; (up to 16 log files)

#### •Enable Web Log

Save the history of web operations (up to 2 log files)

#### •Enable Debug Log

Save debug information (up to 2 log files)

### Packets Capture Tool

This feature always used to capture packets for technician. Integrate packet capture tool "Wireshark" in MyPBX.

Users also could specify the destination IP address and port to get the packets.

#### •IP

Specify the destination IP address to get the packets.

#### •Port

Specify the destination Port to get the packets.

#### •NIC

Choose the NIC (LAN or WAN) which you want to capture the packets.

## 8 Addons



Click **Addons** to enter "Addons" page.

MyPBX supports many addons to enhance the function of MyPBX.

In this page, you can install the according addons, including call recording, MyPBX Client, Bill System and Hotel Module.

Before you install an addon, you should have a license. You can buy the license from Yeastar.

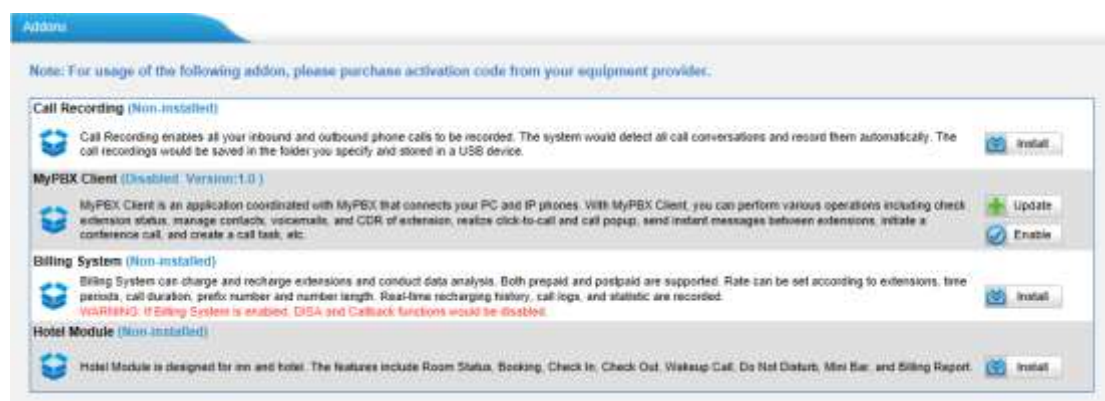


Figure 8-1 Addon Page

### 8.1 Call Recording

After you install "Call Recording" module, you can log in MyPBX using "monitor" to configure recording settings, and store the recording in USB device.

For more details about call recording, please refer to [Appendix H](#).

### 8.2 MyPBX Client

MyPBX Client is an application coordinated with MyPBX that connects your PC and IP phones. With MyPBX Client, you can perform various operations including check extension status, manage contacts, voicemails, and CDR of extension, realize click-to-call and call popup, send instant messages between extensions, initiate a conference call, and create a call task, etc.

For more details about MyPBX client, please refer to the link below:

[http://www.yeastar.com/Solutions/MyPBX\\_Client\\_Solution.asp](http://www.yeastar.com/Solutions/MyPBX_Client_Solution.asp)

## 8.3 Billing System

Billing System caters for the growing need of billing solutions that can efficiently monitor the calls and maintain record of each call received/transferred by IP-PBX. Billing System can charge and recharge extensions and conduct analysis on the basis of statistics. Both prepaid and postpaid are supported. Flexible rate settings and detailed records make it extremely easy for enterprise to monitor charges, spot misuse, and enhance efficiency. With Billing System, call accounting are just clicks away.

**Note:** If Billing System is enabled, DISA and Callback functions would be disabled.

For more details about Bill System, please refer to the link below:

[http://www.yeastar.com/Solutions/Billing\\_System\\_Solution.asp](http://www.yeastar.com/Solutions/Billing_System_Solution.asp)

## 8.4 Hotel Module

MyPBX Hotel Module is designed for inns and hotels. The features include Room Status, Booking, Check In, Check Out, Wakeup Call, Do Not Disturb, Mini Bar, and Billing Report.

For more details about "Hotel Module", please refer to the link below:

[http://www.yeastar.com/Solutions/Hotel\\_Module\\_Solution.asp](http://www.yeastar.com/Solutions/Hotel_Module_Solution.asp)

## 9 Logout



Click  to log out safely..



## 10. Use MyPBX

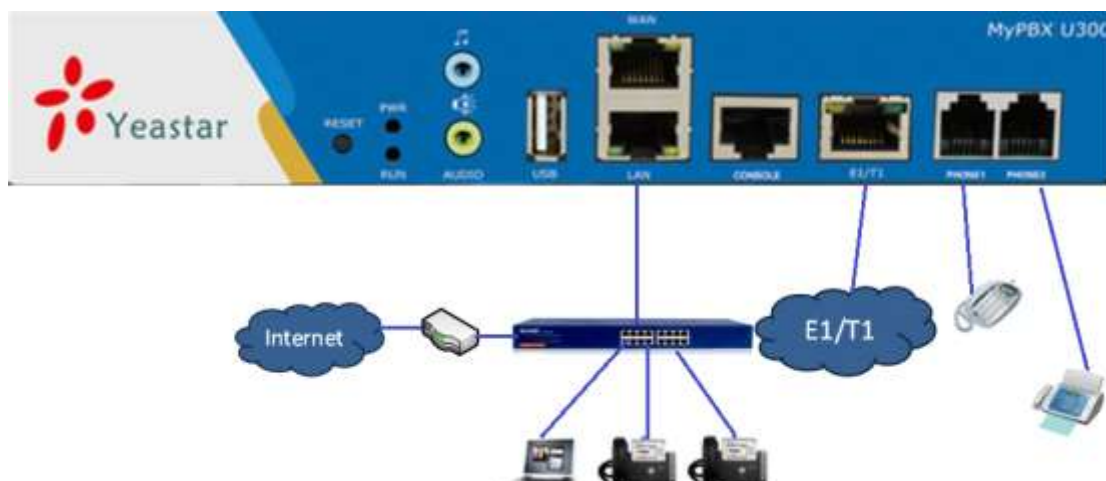


Figure 10-1 U300 Connection Drawing

### 10.1 Make outbound call

To make an outbound call, we need to add trunk first. There are 3 types of VoIP Trunk:

- DigitalTrunk:** E1/T/J1 ports of MyPBX, connected to the ISDN provider.  
You need to get the detailed signaling type from the provider before configure the E1 trunk.
- VoIP Trunk:** Connected to remote VOIP service server.  
You should get an IP address with user name/ password from the provider.
- Service Provider:** Connected to service provider server.  
You will get only IP address for authorization.

#### 10.1.1 Sample Routing via E1 Trunk

Let's route all inside extensions through an E1 trunk by dialing 9. In Outbound Routes, add a new outbound route as below.

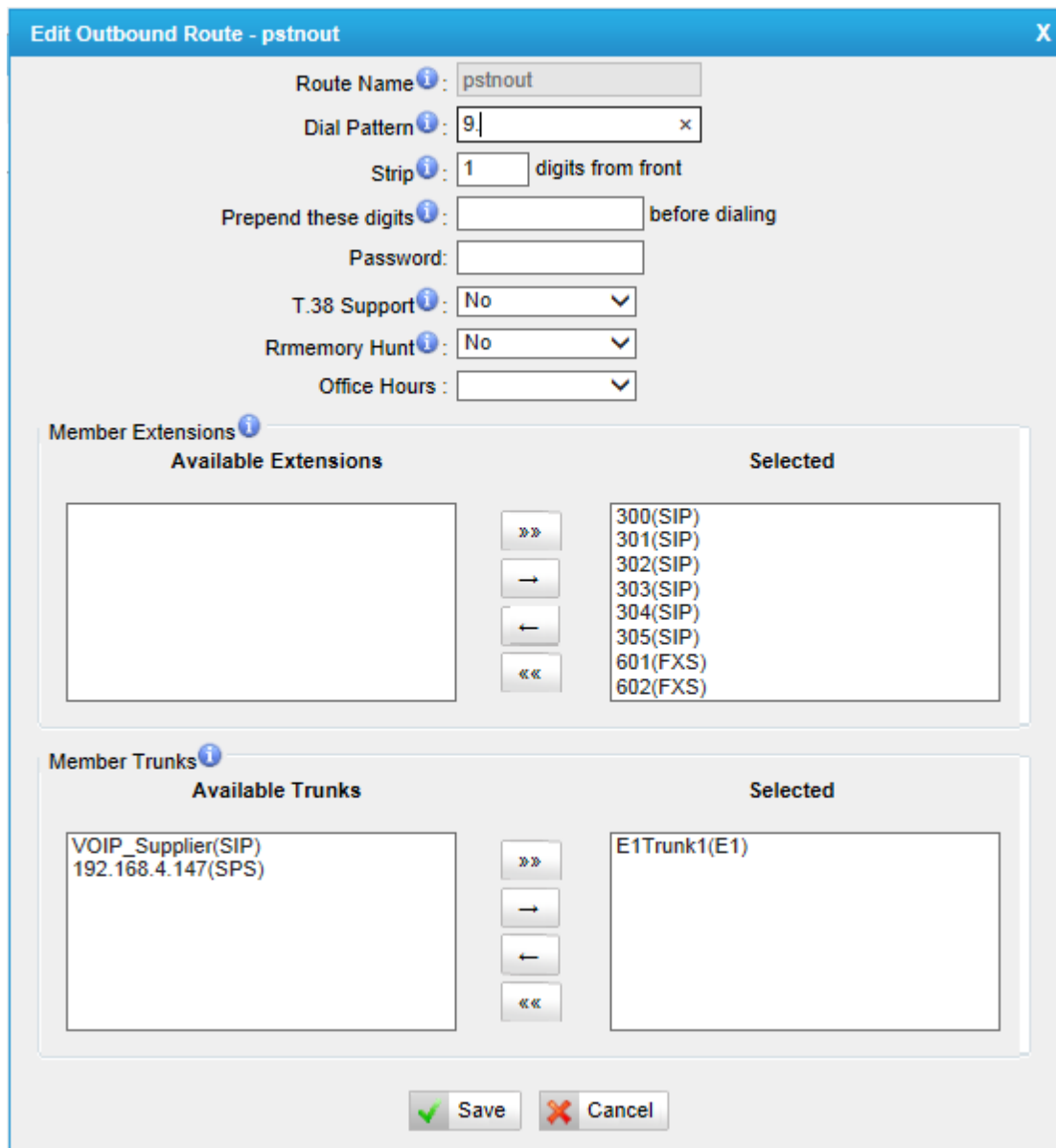


Figure10-2 Create Outbound Route for E1 Trunk

As we can see from the outbound route of 'pstnout', all phone numbers starting with 9 will have their first digit stripped off (digit 9) and will be sent to the E1.

After we have configured the above, we can dial 9 + local number to dial out via anE1 trunk.

**Note:** Setting number prefix to wild card **X**. and setting Strip to **0** digits from the front will allow all calls to go through this outbound route directly.

### 10.1.2 Sample Routing via VoIP Trunk

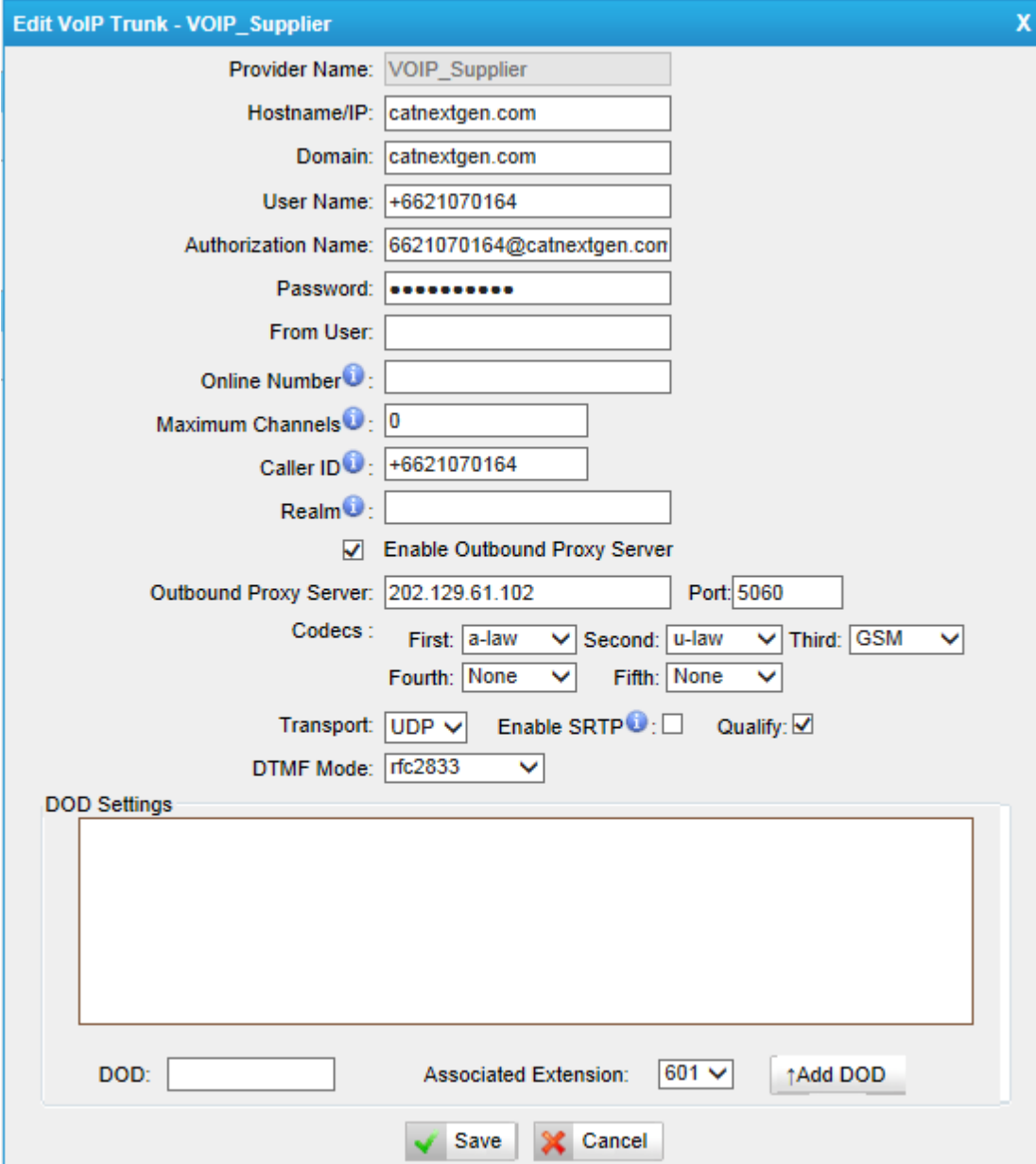
Let's configure all inside extensions to dial '0' through the VoIP Trunk.

## 1. Add VoIP service provider

Before we do add this, please make sure you have a VoIP Trunk account.

Trunks→VoIP Trunk →SIP Trunk

Enter your account information on this page, and click Save.



**Edit VoIP Trunk - VOIP\_Supplier**

Provider Name: VOIP\_Supplier

Hostname/IP: catnextgen.com

Domain: catnextgen.com

User Name: +6621070164

Authorization Name: 6621070164@catnextgen.com

Password: .....

From User:

Online Number:

Maximum Channels: 0

Caller ID: +6621070164

Realm:

☒ Enable Outbound Proxy Server

Outbound Proxy Server: 202.129.61.102 Port: 5060

Codecs: First: a-law Second: u-law Third: GSM Fourth: None Fifth: None

Transport: UDP Enable SRTP: ☐ Qualify: ☒

DTMF Mode: rfc2833

**DOD Settings**

DOD: Associated Extension: 601

Figure10-3 Create an VOIP Trunk

## 2. Add Outbound Routes

As we can see from the Outbound Route of 'VOIP\_OUT', all phone numbers starting with 0 will have their first digit stripped off (digit 0) and will be sent to the SIP Trunk.

Edit Outbound Route - VOIP\_OUT

Route Name: VOIP\_OUT  
Dial Pattern: 0  
Strip: digits from front  
Prepend these digits: before dialing  
Password:  
T.38 Support: No  
Rmemory Hunt: No  
Office Hours: default

Member Extensions

Available Extensions

Selected

300(SIP)  
301(SIP)  
302(SIP)  
303(SIP)  
304(SIP)  
305(SIP)  
601(FXS)  
602(FXS)

Member Trunks

Available Trunks

Selected

E1Trunk1(E1)  
192.168.4.147(SPS)

VOIP\_Supplier(SIP)

Save Cancel

Figure10-4 Create an Outbound route for VOIP trunk

Now that we have added two outbound dialing rules, any call starting with 9 will be routed to the E1, and any number starting with 0 will be routed to the SIP Trunk.

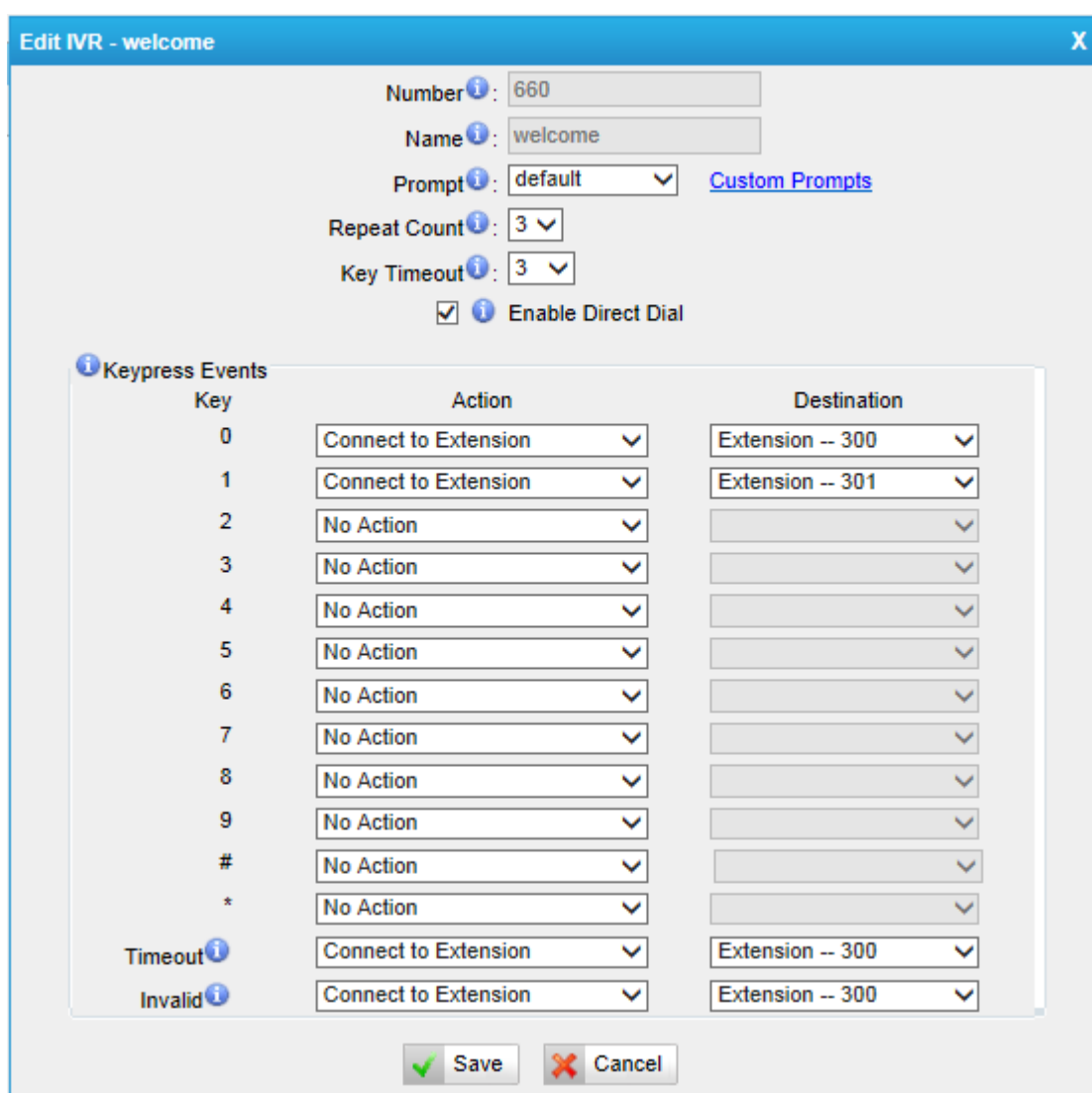
## 10.2 Incoming call

### 10.2.1 Sample Routing to an IVR

Let's configure an incoming call to route to the IVR. In the IVR itself, let's configure digit 0 to route the call to extension 500, and digit 1 to route the call to extension 501.

#### 1. Add IVR

To add a new IVR, go to IVR→ Create New IVR



**Edit IVR - welcome**

Number: 660

Name: welcome

Prompt: default [Custom Prompts](#)

Repeat Count: 3

Key Timeout: 3

☒ Enable Direct Dial

Key	Action	Destination
0	Connect to Extension	Extension -- 300
1	Connect to Extension	Extension -- 301
2	No Action	
3	No Action	
4	No Action	
5	No Action	
6	No Action	
7	No Action	
8	No Action	
9	No Action	
#	No Action	
*	No Action	
Timeout	Connect to Extension	Extension -- 300
Invalid	Connect to Extension	Extension -- 300

Figure10-5 Create IVR

#### 2. Add Inbound Routes

As we can see from the Inbound Route of 'VOIP\_IN', all incoming calls from SIP trunk will be sent to the IVR.

Edit Inbound Route: VOIP\_IN
X

**General**

Route Name ? :   
DID Number ? :   
Extension ? :   
Caller ID Number ? :   
Distinctive Ringtone ? :   
Enable Callback : No v [Callback Settings](#)

**Member Trunks** ?

**Available Trunks**

E1Trunk1(E1)  
192.168.4.147(SPS)

»»  
→  
←  
««

**Selected**

VOIP\_Supplier(SIP)

**Business Days**

Office Hours : default v  
Office Hours Destination : IVR v IVR -- welcome v  
Non-office Hours Destination : End Call v  v

**During Holidays**

Holiday :  v  
Destination : End Call v  v

**Fax Detection**

Destination : No Detect v  v

✔ Save

✖ Cancel

Figure10-6 Create Inbound Route for VOIP Trunk

## APPENDIX A FAQ

### Q1. How to Register SIP devices?

**A1:**

1) Register SIP softphone

Download the x-lite softphone from counterpath website

[www.counterpath.com](http://www.counterpath.com)

After install the x-lite, right click the panel and select the SIP Account setting and then configure it.

**Display Name:** 500

**User Name:** 500

**Password:** 500

**Authorization Name:** 500

**Domain:** 192.168.5.150

2) Register IP Phone (for example, Yealink's T28 IP Phone)

a) Connect the T28's Internet port to the switch. And it can get the IP from your route.

b) Press the 'OK' key on T28 to get the IP of T28.

c) Put the IP on web browser then you can enter the T28 configure page through this IP.

d) Put the SIP extensions info on the T28 IP phones.

**Display Name:** 501

**User Name:** 501

**Register Name:** 501

**Password:** 501

**SIP Server:** 192.168.5.150

Use the same method register another T28 to other extension.

### Q2. How do I reset MyPBX back to the factory default settings?

**A2:** To perform a reset, please follow steps below:

**Step 1:** Hold down the 'Reset' button on the back of the unit for 5 seconds and watch the LEDs on the front of the MyPBX. When the status LED turns red, let go of the reset button.

**Step 2:** When the RUN status LED starts blinking, MyPBX will be set back to factory defaults.

**Step 3:** To access the configuration page, navigate to **192.168.5.150** using a web browser. Make sure that you are on the 192.168.5.0 subnet before doing this.

**Step 4:** Login to the device with the username '**admin**' and the password '**password**', in order to begin reconfiguring the device.

## APPENDIX B How to Configure Autobackup

Before Autobackup can be properly configured, an SMB share folder accessible from MyPBX must be set up on a Windows based machine. Once that has been set up, please follow the steps below.

**Step 1** Add a new folder, rename it, and set this new folder's share Properties according to Figure B-1

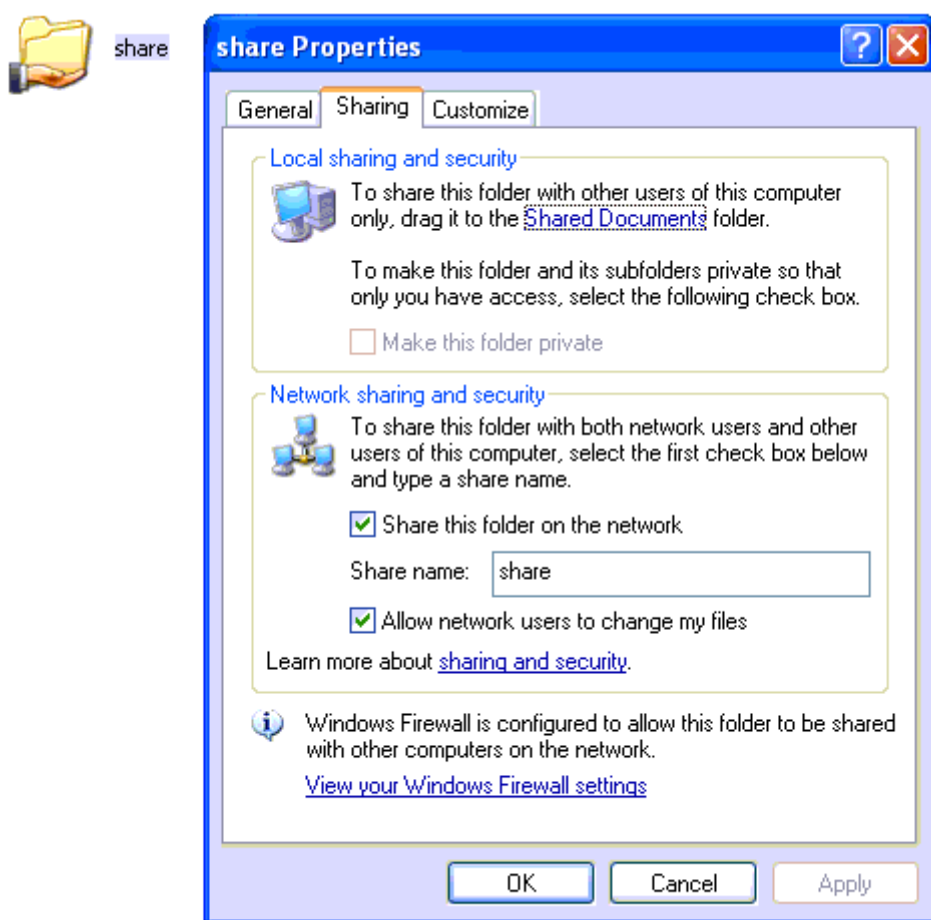


Figure B-1 Set up share Properties

**Step 2** Enter the new folder and create a new text file, then rename this file to status.txt. This step is very important, DO NOT forget to create the status.txt file.

**Step 3** Configure Autobackup settings on MyPBX to Figure B-2



**External Storage Settings**

The External Storage feature is used to extend storage space. Once configured, the files(voicemail, call recording files, CDR files) created before the configured days will be moved to the Net-Disk.

**Step 1: Create a Net-Disk on a chosen computer**

**Step 2: Input the Net-Disk properties**

Net-Disk Host/IP:

Net-Disk Share Name:

Net-Disk Access Username:

Net-Disk Access Password:

Move files created before:  days ago

**Step 3: Save Net-Disk settings**

**Step 4: Make sure the settings are successfully completed**

Figure B-2 Autobackup Setting

**Net-Disk Host/IP:** Change this to the IP address of the computer where backup files will be stored.

**Net-DiskShareName:** Change this to the name of the shared folder where backups will be stored.

**Net-DiskShareUsername:** The user name used to log into the network share. Leave this blank if it is not required

**Net-DiskSharePassword:** The password used to log into the network share. Leave this blank if it is not required

If configuring is correctly, open your Windows share folder to see if the MyPBX backup files and folders has been created. If the contents of the backup folder look similar to Figure B-3, then you have successfully configured Autobackup on the MyPBX unit.

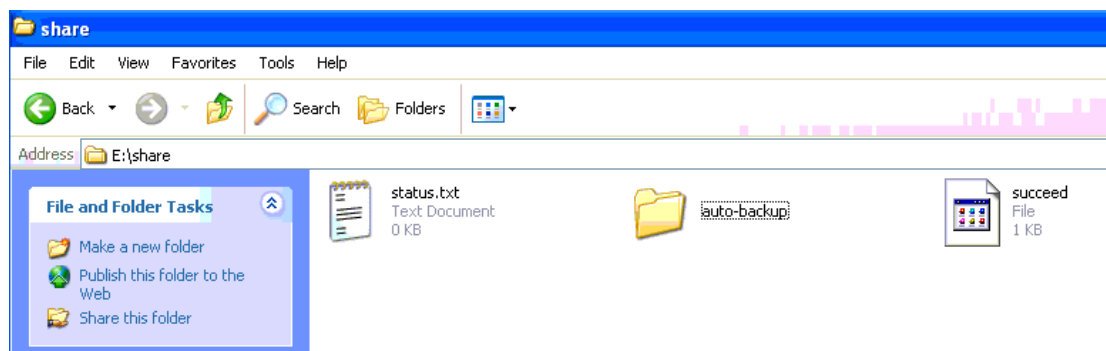


Figure B-3 Autobackup setting succeed

# APPENDIX C How to Configure NAT setting

**When MyPBX is behind a NAT(firewall),you need to configure NAT setting on MyPBX if you want to use a remote extension.**

Please follow section **1** or **2** below depending on your network configuration.

**1.** If MyPBX is connected to a local network, you must set up port forwarding on your router. Specifically, you must map port 5060 (default SIP port) and port 10001-10200(default RTP port range) as UDP ports.

Next, log in MyPBX WebGUI, go to 'PBX'-'>'Advanced Settings' ->'SIP Settings' ->'NAT'

**External IP Address:** your router's public IP address

**External Host:** your router's domain

**ExternalRefresh Interval:** 20 seconds

**Local Network Identification:**192.168.5.0/255.255.255.0 (change this according to your network configuration)

**NAT mode:** Yes

**Allow RTP Reinvite:** Yes

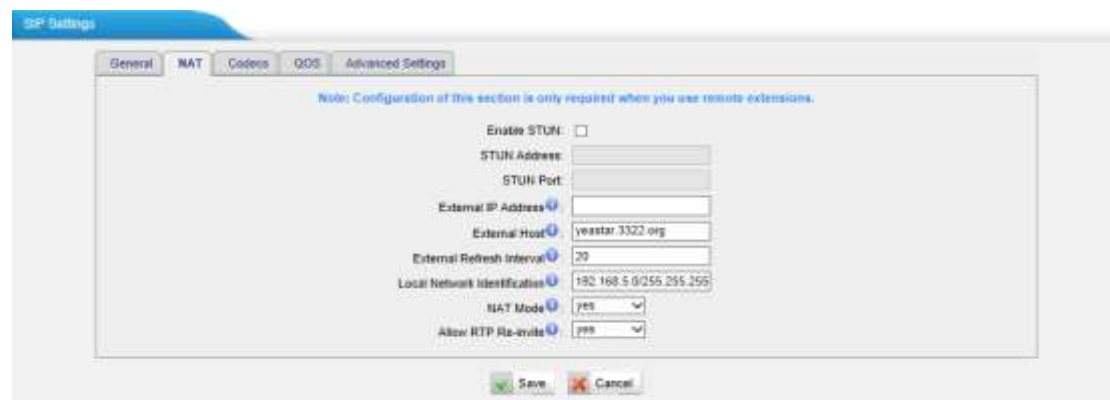


Figure C-1 NAT Settings

Assuming that your router's host address is yeastar.3322.org, your local network is from 192.168.5.1-192.168.5.254, and the subnet Mask is 255.255.255.0, the MyPBX network settings should configured like Figure C-2

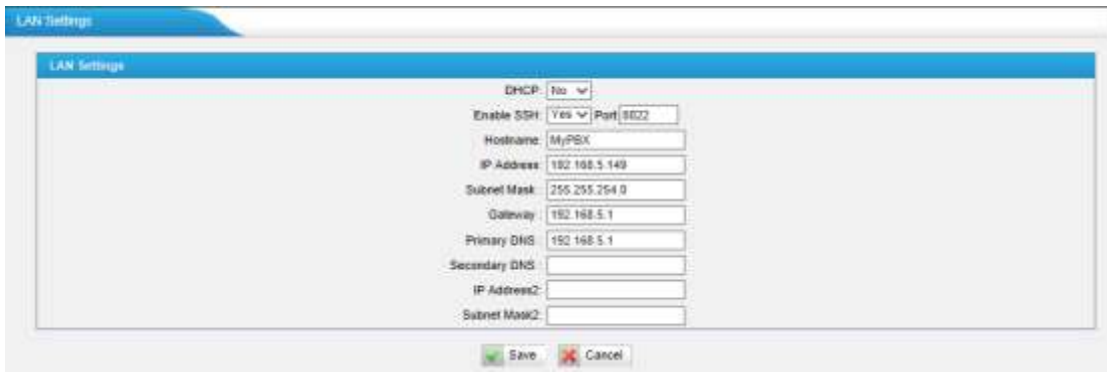


Figure C-2 MyPBX Network setting

**2.** If MyPBX has a public IP, (i.e. is connected directly to your internet service provider), the network settings should be configured according to Figure C-3:

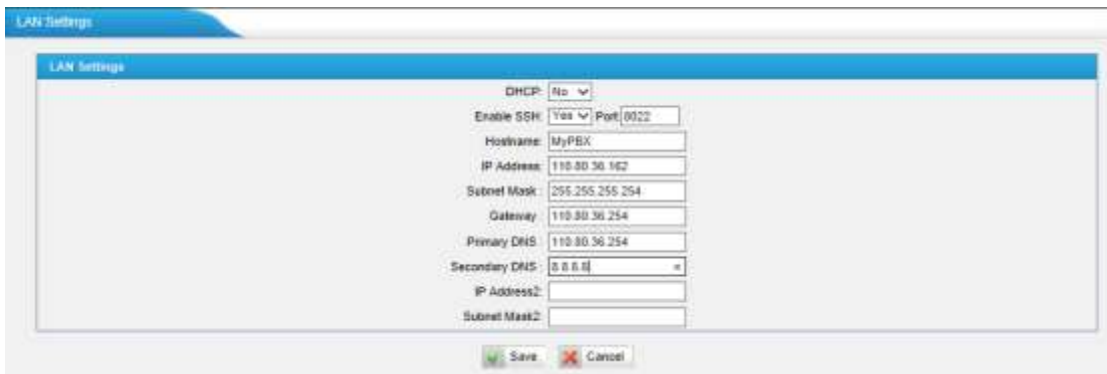


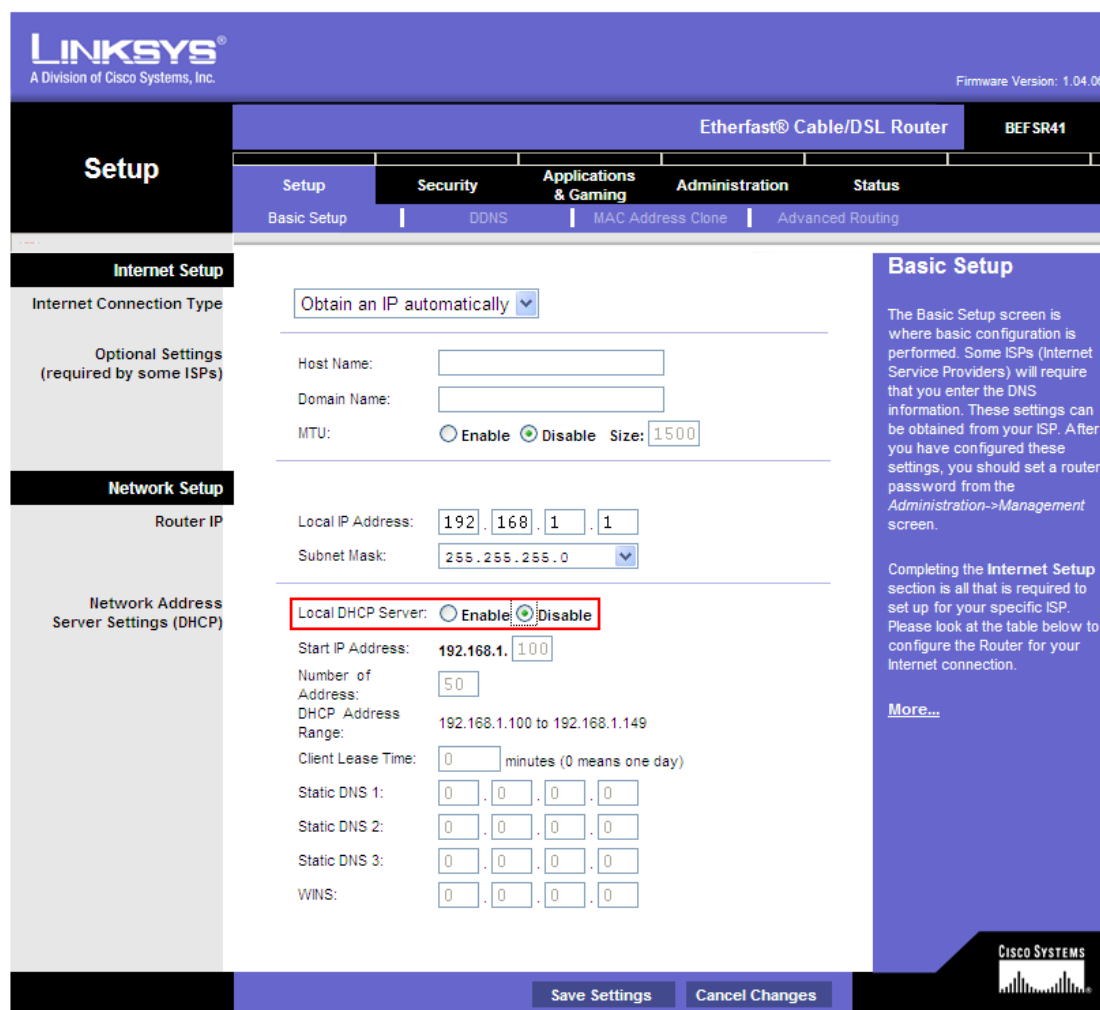
Figure C-3

Now, MyPBX has been configured as a public IP, so there is no need to configure NAT again, just leave all settings in 'NAT' blank.

# APPENDIX D How to Use Auto Provision

**Step1.** Disable DHCP Server on your local network.

E.g. Disable DHCP Server on Linksys Router.



The screenshot shows the Linksys Basic Setup page for an Etherfast Cable/DSL Router (BEFSR41). The 'Local DHCP Server' is set to 'Disable'. The 'Internet Setup' section shows 'Obtain an IP automatically' selected. The 'Network Setup' section shows the 'Local IP Address' as 192.168.1.1 and the 'Subnet Mask' as 255.255.255.0. The 'Local DHCP Server' section shows the 'Local DHCP Server' set to 'Disable' (indicated by a red box around the 'Disable' radio button). The 'Start IP Address' is 192.168.1.100, the 'Number of Address' is 50, and the 'DHCP Address Range' is 192.168.1.100 to 192.168.1.149. The 'Client Lease Time' is 0 minutes (0 means one day). The 'Static DNS' and 'WINS' sections are empty. The 'Save Settings' and 'Cancel Changes' buttons are at the bottom.

Figure D-1 Disable DHCP Server on Router

**Step2.** Enable DHCP Server on MyPBX.

Login MyPBX web interface, 'System'→'Network Preference' ->'DHCP Server'→'Enable DHCP Server'.



Figure D-2 Enable DHCP Server on MyPBX

**Step3.** Configure phones on MyPBX auto-provision page.

1. Login MyPBX web interface, 'PBX' -> 'Extensions' -> 'Phone Provisioning' -> 'Add Phone'.

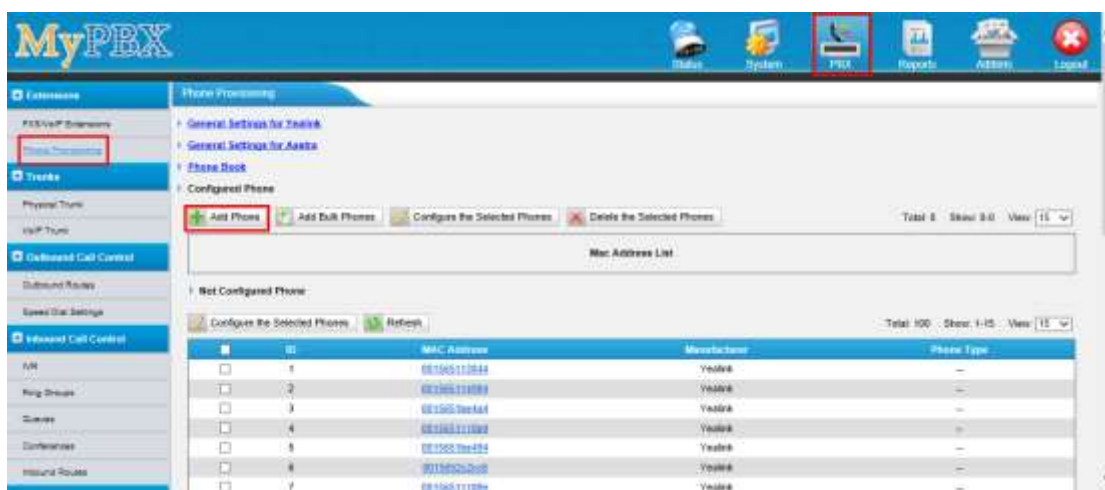


Figure D-3 Add Phone

2. Fill in the phone detail message on the pop-up windows.

Input IP Phone's MAC address, configure Name, Call waiting, Line, Extension, Label, Line active for the phone. And also you can configure other features on the phone, like codecs, memory keys etc.

Add Phone
X

General
Codecs
Memory Key Settings
Line Keys Settings

Enabled: Yes
NewConfig: Yes

MAC Address: 001565
Name:

Manufacturer: Yealink
Phone Type: T28

Call Waiting: Enabled
Key As Send: #

Auto Redial: Disabled
Auto Answer: Disabled

Phone Book: Enabled

Line

☐ Line1
Extension:
Label:
Line Active:

☐ Line2
Extension:
Label:
Line Active:

☐ Line3
Extension:
Label:
Line Active:

☐ Line4
Extension:
Label:
Line Active:

☐ Line5
Extension:
Label:
Line Active:

☐ Line6
Extension:
Label:
Line Active:

Save
Cancel

Figure D-4 Yealink T28 Phone Provisioning

**Step4.** Turn on the power and connect the network cable to IP Phone.

**Remark:** The factory default setting of DHCP for IP Phone is enabled, so you can skip this step to step 5.

If the DHCP is disabled, please follow below step to enable it. (e.g. Yealink's IP Phone).

1. Login IP phone's web page.
2. Enable DHCP.



The screenshot shows the Yealink U300 Administrator's Guide interface. The top navigation bar includes tabs for Status, Account, Network, Phone, Contacts, Upgrade, and Security. The Network tab is selected, and the sub-tab is Internet Port (WAN). The DHCP option is selected and highlighted with a red box. The Static IP Address and PPPoE options are also visible. A NOTE section on the right explains the DHCP, Static IP Address, and PPPoE settings.

**NOTE**

**DHCP**  
The device will acquire its IP address from the DHCP server automatically.

**Static IP Address**  
Set the IP address, Subnet Mask, Default Router IP address, Primary DNS, Secondary DNS fields manually.

**PPPoE**  
This setting provide by DSL.

Figure D-5 Enable DHCP on IP phone

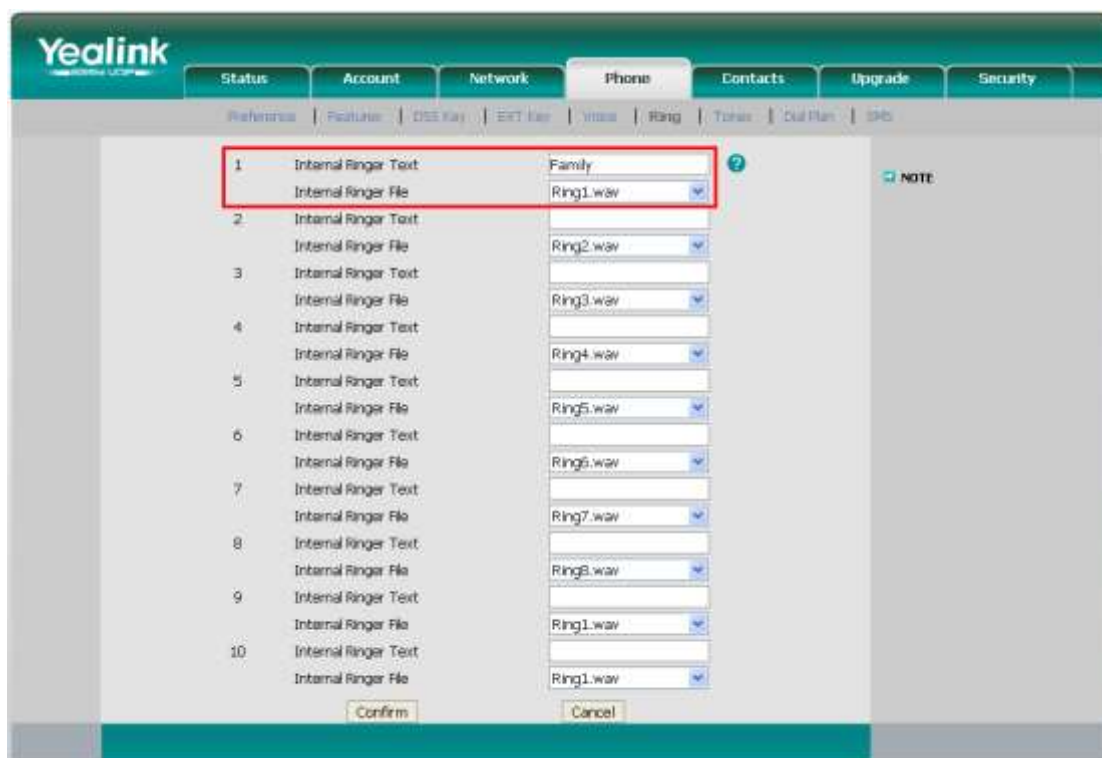
**Step5.** Finish.

## APPENDIX E How Do I Configure Distinctive Ring Tones

**Step1:** On your IP phone, navigate to the Phone settings web configuration page and find the Distinctive Ring Tone section.

For each custom ring tone, enter the Internal Ringer Text (can be digits or text) to trigger the ring tone. For example, you may enter "Family".

e.g.: Yealink's IP phone.



FigureE-1 Configure Ring Text on IP Phone

**Step2.** Configure the 'Distinctive Ringtone' on MyPBX.

Login MyPBX web interface, 'PBX' -> 'Inbound Call Control' -> 'Inbound Routes' -> Edit Inbound Route, fill in the Internal Ringer Text on 'Distinctive Ringtone'.



Edit Inbound Route: VOIP\_IN
X

General

Route Name : VOIP\_IN
DID Number :
Extension :
Caller ID Number :
Distinctive Ringtone : family
Enable Callback : No [Callback Settings](#)

Member Trunks

Available Trunks

E1Trunk1(E1)  
192.168.4.147(SPS)

»»  
→  
←  
««

Selected

VOIP\_Supplier(SIP)

Business Days

Office Hours : default
Office Hours Destination : IVR IVR -- welcome
Non-office Hours Destination : End Call

During Holidays

Holiday :
Destination : End Call

Fax Detection

Destination : No Detect

Save Cancel

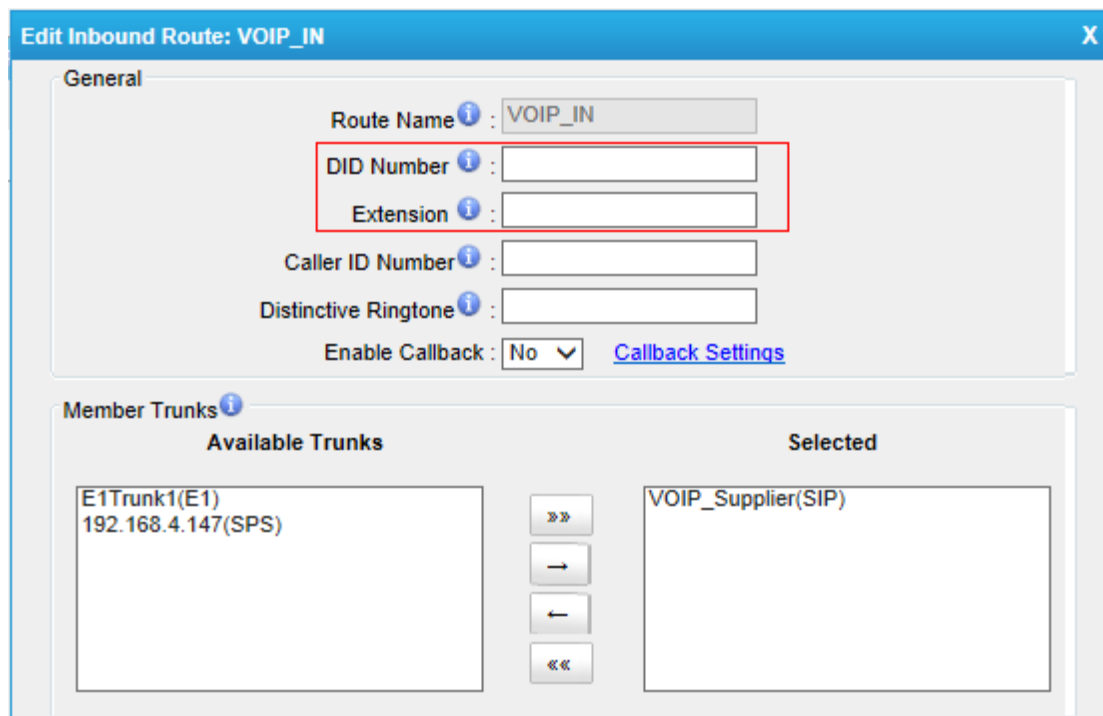
FigureE-2 Configure Distinctive in Inbound Route

**Step3.** Finish.

## APPENDIX F How to Use DID

Direct inward dialing (DID), also called direct dial-in (DDI) in Europe and Oceania, is a feature offered by telephone companies for use with their customers' private branch exchange (PBX) systems. In DID service the telephone company provides one or more trunk lines to the customer for connection to the customer's PBX and allocates a range of telephone numbers to this line (or group of lines) and forwards all calls to such numbers via the trunk.

MyPBX support DID, you can configure DID in inbound route. Related settings: **DID Number, Extension, Destination.**



The screenshot shows the 'Edit Inbound Route: VOIP\_IN' window. The 'General' tab is selected. The 'Route Name' is 'VOIP\_IN'. The 'DID Number' and 'Extension' fields are highlighted with a red box. The 'Caller ID Number' and 'Distinctive Ringtone' fields are empty. The 'Enable Callback' is set to 'No'. The 'Member Trunks' section shows 'Available Trunks' (E1Trunk1(E1), 192.168.4.147(SPS)) and 'Selected' (VOIP\_Supplier(SIP)).

Figure F-1 Inbound Route

### •DID Number

Define the expected DID Number if this trunk passes DID on incoming calls. Leave this field blank to match calls with any or no DID info. Only service provider, E1 trunks, BRI trunks or SIP trunks need to be configured with this setting.

You can also use pattern matching to match a range of numbers. The following patterns may be used:

**X:** Any Digit from 0-9

**Z:** Any Digit from 1-9

**N:** Any Digit from 2-9

**[12345-9]:** Any digit in the brackets (in this example, 1,2,3,4,5,6,7,8,9)

The '.'Character will match any remaining digits. For example, 9011. will match any phone number that starts with 9011, excluding 9011 itself.

The '!' will match none remaining digits, and causes the matching process to complete as soon as it can be determined that no other matches are possible.

**•Extension**

Define the extension for DID number, this field only valid when use E1 trunk for this inbound router. You can only input number and '-' in this field, and the format can be xxx or xxx-xxx. The count of the number must be only one or equal the count of the DID number.

**•Destination**

If you don't set the extension, you can set the destination of the call here.

**Example 1:**

Step1: You set the DID number (5503XXX in this example).

Step2: You choose the destination(IVR in this example).

The configuration of this example means when the incoming call with DID number 5503XXX (7 digits number start with 5503) will go to the destination IVR.

If you choose the destination, please leave the Extension form blank.

Edit Inbound Route: pstnin

General

Route Name : pstnin  
DID Number : 5503XXX  
Extension :  
Caller ID Number :  
Distinctive Ringtone :  
Enable Callback : No [Callback Settings](#)

Member Trunks

Available Trunks

Selected

VOIP\_Supplier(SIP)  
192.168.4.147(SPS)

»  
→  
←  
«

E1Trunk1(E1)

Business Days

Office Hours : default  
Office Hours Destination : IVR IVR -- welcome  
Non-office Hours Destination : IVR IVR -- welcome

During Holidays

Holiday :  
Destination : End Call

Fax Detection

Destination : No Detect

Save Cancel

Figure F-2 Configure DID in Inbound Route

### Example 2:

Step1: You set the DID number (6001-6099 in this example).

Step2: You set the Extension (6001-6099 in this example).

The configuration of this example means when the incoming call with DID number 6001 to 6099 will go to the destination 6001 to 6099(number 6001 to extension 6001, number 6002 to extension 6002).

The destination you set below will be disabled if you set the Extension.

Edit Inbound Route: VOIP\_IN
X

General

Route Name : VOIP\_IN  
DID Number : 6001-6099  
Extension : 6001-6099  
Caller ID Number :  
Distinctive Ringtone :  
Enable Callback : No [Callback Settings](#)

Member Trunks

Available Trunks		Selected
E1Trunk1(E1) 192.168.4.147(SPS)	»» → ← ««	VOIP_Supplier(SIP)

Business Days

Office Hours : default  
Office Hours Destination : IVR IVR -- welcome  
Non-office Hours Destination : End Call

During Holidays

Holiday :  
Destination : End Call

Fax Detection

Destination : No Detect

Save Cancel

Figure F-3 Configure DID and Extension in Inbound Route

# APPENDIX G MyPBX Security configuration Guide

VoIP attacks, although it is not an everyday occurrence, it does exist. While using VoIP, system security is undoubtedly one of the issues we care about most. But with the appropriate configuration, and some basic safety habits, we can improve the security of the telephone system. Moreover, the powerful built-in firewall function in MyPBX is adequate to enable the system to run safely and stably.

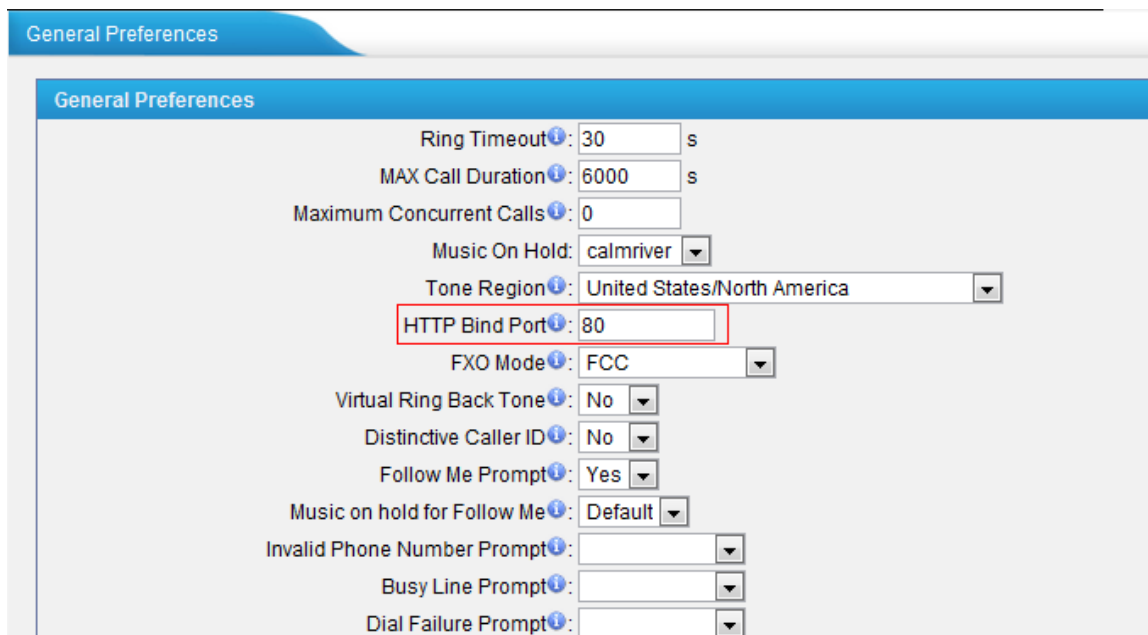
This guide will introduce the highest defense level in MyPBX, and we strongly recommend that you configure firewall and other security options according to this guide, to prevent the attack fraud and the system failure or calls loss.

**Note:** In this guide, the configuration options marked with '\*' only exist in 12.17.XX.XX and above versions, namely, 3.2 guest calls option, 3.3 remote registered option, and 5 alarm settings.

## 1. Security Configuration for Web GUI

### 1.1 Change the default access port for HTTP on Options page

PBX→Basic Settings→General Preferences→HTTP Bind Port



General Preferences

General Preferences

Ring Timeout: 30 s

MAX Call Duration: 6000 s

Maximum Concurrent Calls: 0

Music On Hold: calmrivier

Tone Region: United States/North America

HTTP Bind Port: 80

FXO Mode: FCC

Virtual Ring Back Tone: No

Distinctive Caller ID: No

Follow Me Prompt: Yes

Music on hold for Follow Me: Default

Invalid Phone Number Prompt:

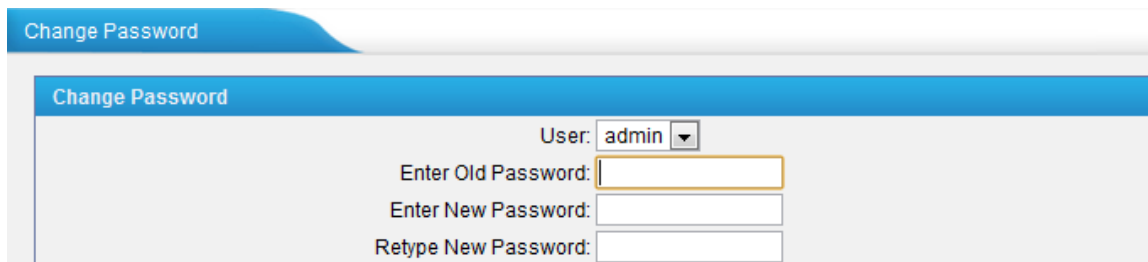
Busy Line Prompt:

Dial Failure Prompt:

Figure G-1 Modifying HTTP port in General Preference

## 1.2 Change the default password for the web GUI

System→System Preferences→Change Password



Change Password

Change Password

User: admin

Enter Old Password:

Enter New Password:

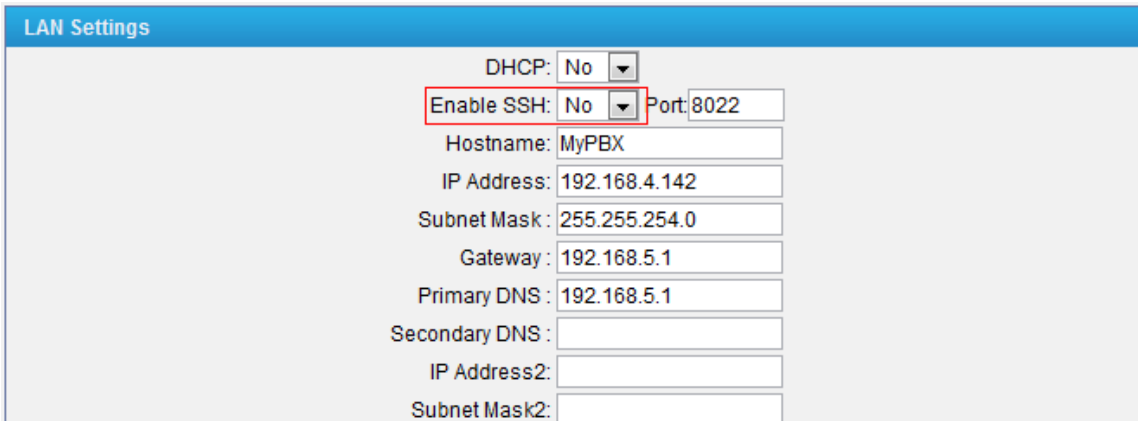
Retype New Password:

Figure G-2 Modifying the Password for Web GUI

## 2. Disable SSH on LAN Settings Page

### 2.1 Disable SSH

Select LAN Settings→Enable SSH. If external debugging isn't required, please select "No".



LAN Settings

DHCP: No

Enable SSH: No Port: 8022

Hostname: MyPBX

IP Address: 192.168.4.142

Subnet Mask: 255.255.254.0

Gateway: 192.168.5.1

Primary DNS: 192.168.5.1

Secondary DNS:

IP Address2:

Subnet Mask2:

Figure G-3 Disable SSH

## 2.2 Change the default password for SSH

We can use the Linux command **passwd** to change root password of MyPBX.

1. Login via putty.exe

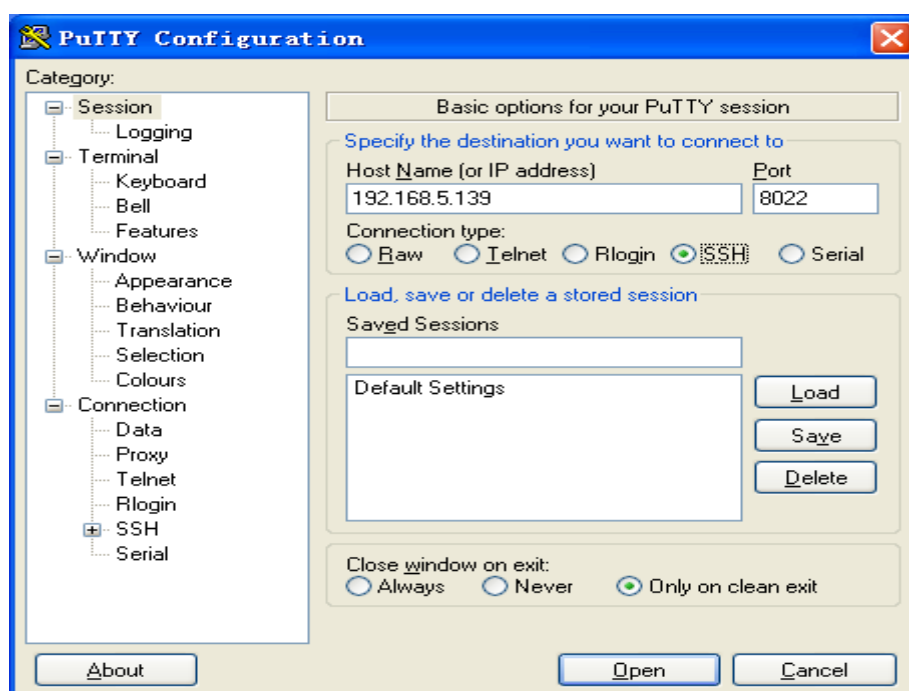


Figure G-4 Log in MyPBX SSH via Putty

2. The default username is **root** and the default password is **ys123456**.



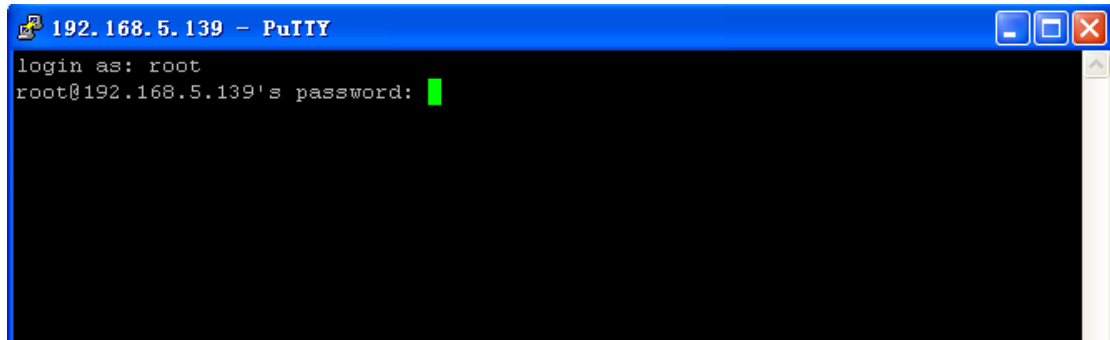


Figure G-5 Enter SSH Username and Password

3. Use command **passwd** to change the root's password

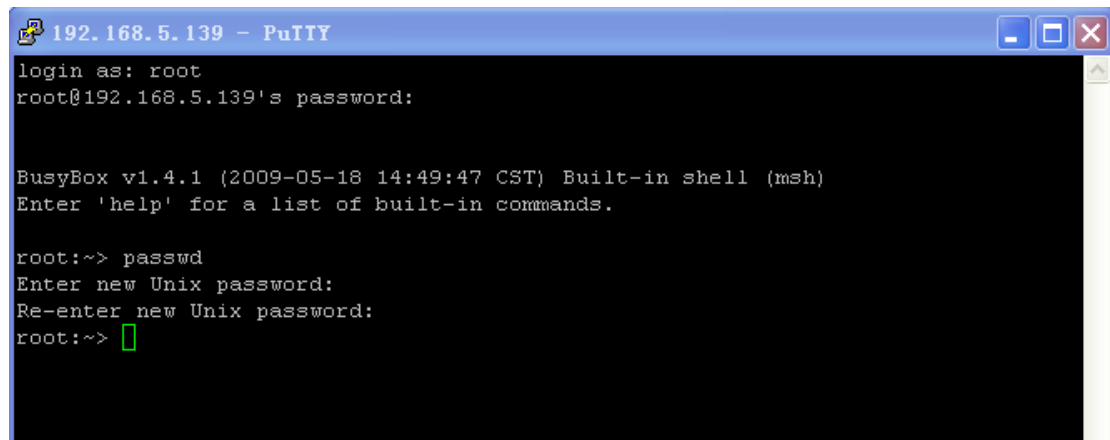
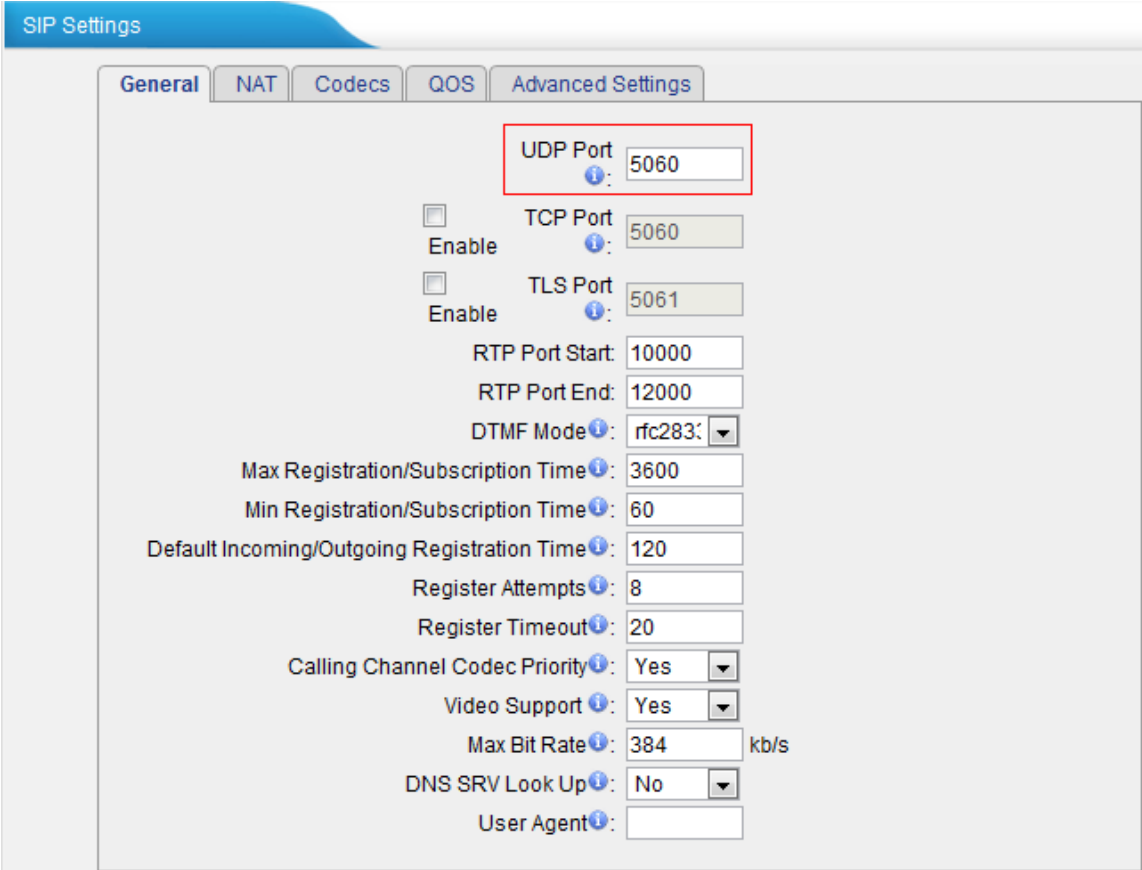


Figure G-6 Modify SSH Password

## 3. Security Configuration for Extensions

### 3.1 Change the default SIP Port

PBX→Basic settings→SIP Settings→General→UDP Port

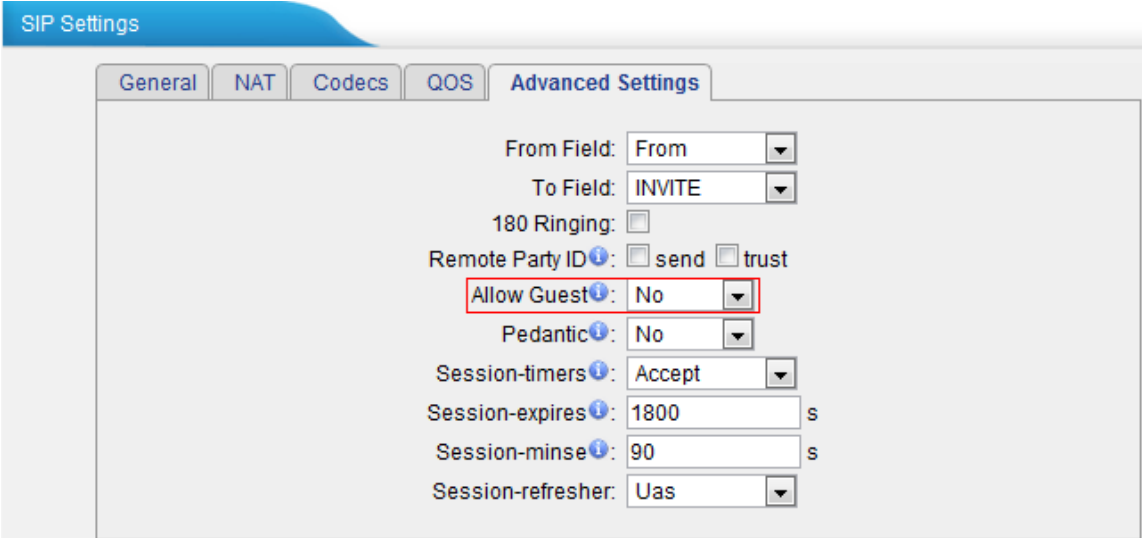


The image shows the 'SIP Settings' window with the 'Advanced Settings' tab selected. The 'UDP Port' field is highlighted with a red box and contains the value '5060'. Other visible settings include: 'TCP Port' (5060), 'TLS Port' (5061), 'RTP Port Start' (10000), 'RTP Port End' (12000), 'DTMF Mode' (rfc283), 'Max Registration/Subscription Time' (3600), 'Min Registration/Subscription Time' (60), 'Default Incoming/Outgoing Registration Time' (120), 'Register Attempts' (8), 'Register Timeout' (20), 'Calling Channel Codec Priority' (Yes), 'Video Support' (Yes), 'Max Bit Rate' (384 kb/s), 'DNS SRV Look Up' (No), and 'User Agent' (empty).

Figure G-7 Modify SIP port

## 3.2 Disable guest calls

PBX→Basic settings→SIP Settings→Advanced Settings→Allow Guest



The image shows the 'SIP Settings' window with the 'Advanced Settings' tab selected. The 'Allow Guest' dropdown menu is highlighted with a red box and set to 'No'. Other visible settings include: 'From Field' (From), 'To Field' (INVITE), '180 Ringing' (unchecked), 'Remote Party ID' (send/trust), 'Pedantic' (No), 'Session-timers' (Accept), 'Session-expires' (1800 s), 'Session-minse' (90 s), and 'Session-refresher' (Uas).

Figure G-8 Don't Allow Guest

### 3.3 Security Configuration for remote extensions

If remote registration isn't required, please disable it.

PBX→Extensions→FXS/VoIP Extensions→VoIP Extensions→General→VoIP Settings

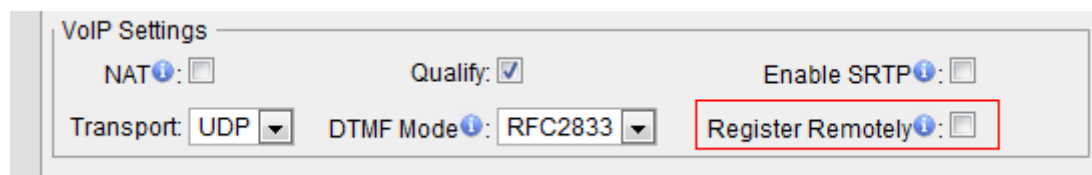


Figure G-9 Register Remotely

### 3.4 Set a robust password and enable IP restriction for extensions

1) Set a new extension password at a higher security level, e.g. AjK5Up1G.

PBX→Extensions→FXS/VoIP Extensions→VoIP Extensions→General→Password

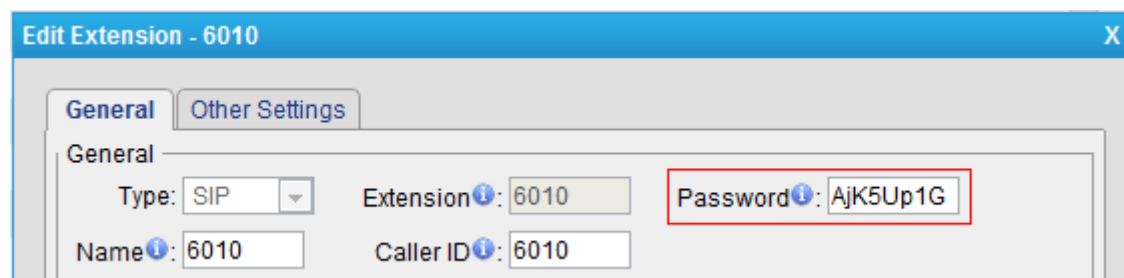


Figure G-10 Modify Extension Password

2) Enable IP restriction and enter the permitted "IP address/Subnet mask", e.g. 192.168.5.136.

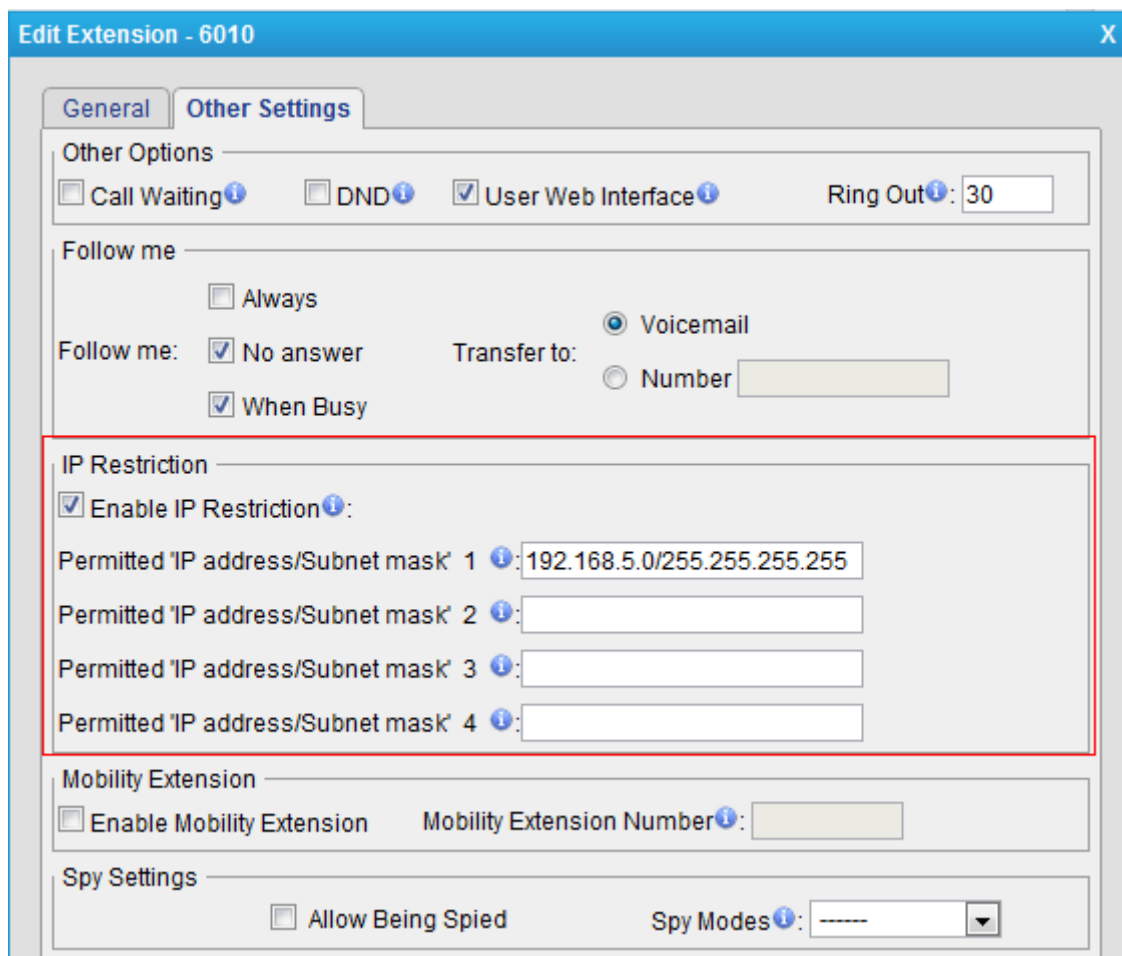


Figure G-11 Enable IP Restriction

## 4. Setup Proper Firewall Rules

**Note:** Please backup the configurations on backup and restore page before you go ahead. In the case that you lock the device, you can reset to factory default and restore the previous configurations. The example rules below works with MyPBX firmware version 2.15.xx.xx or higher versions.

**Step1.** Enable firewall on firewall page of MyPBX.

System→Firewall Settings→Firewall Rules→General Settings

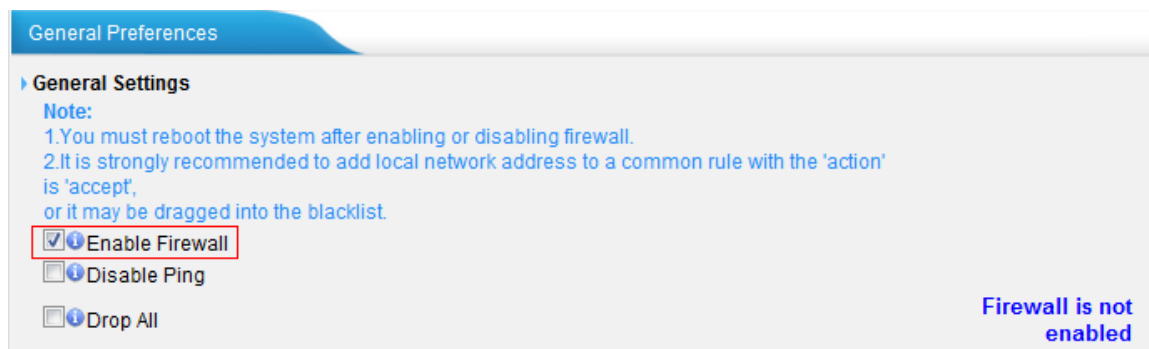


Figure G-12 Enable Firewall

**Step2.** Add a common rule to accept local network access.

Create a common rule to allow the all the IP addresses of the local phones to access MyPBX server. For example, if the IP addresses of the local network are 192.168.5.1-254, the configurations could be as below:

**Name:** LocalNetwork

**Protocol:** BOTH

**Port:** 1:65535

**IP:** 192.168.5.0/255.255.255.0

**Action:** Accept

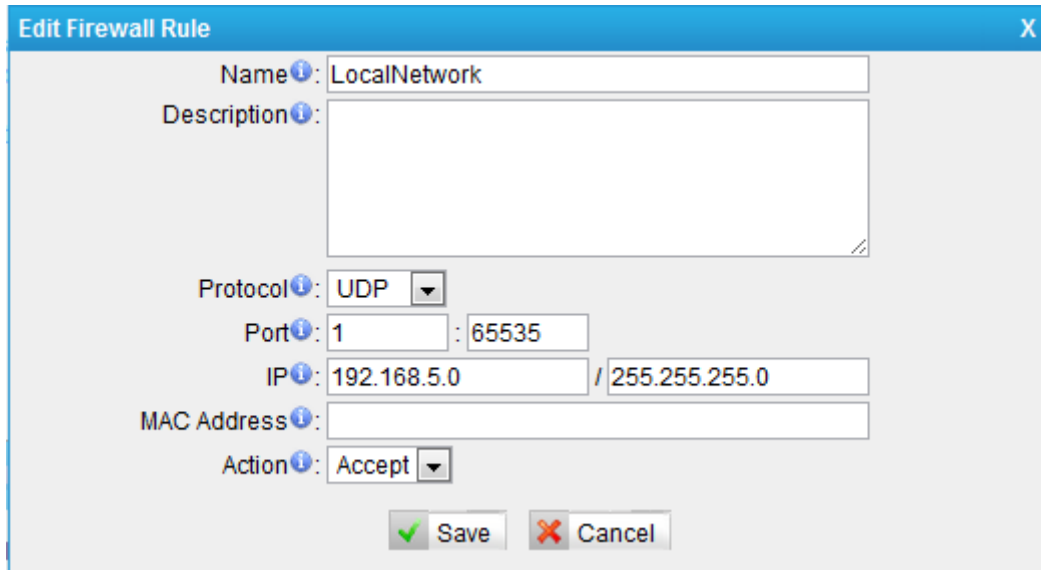


Figure G-13 Add Local Network in Whitelist

**Step3.** Create common rules to accept remote extensions or remote administrators, if you use SIP trunk, please accept the provider's host as well.

**Note:** If there are no remote extensions, the rule is not required.

- 1) Set up the firewall rule to allow the public IP address of remote extensions to access MyPBX server. e.g.110.111.132.6, the configurations could be as below:

**Name:** Remote Extension

**Protocol:** BOTH

**Port:** 1:65535

**IP:** 110.111.132.6/255.255.255.255

**Action:** Accept

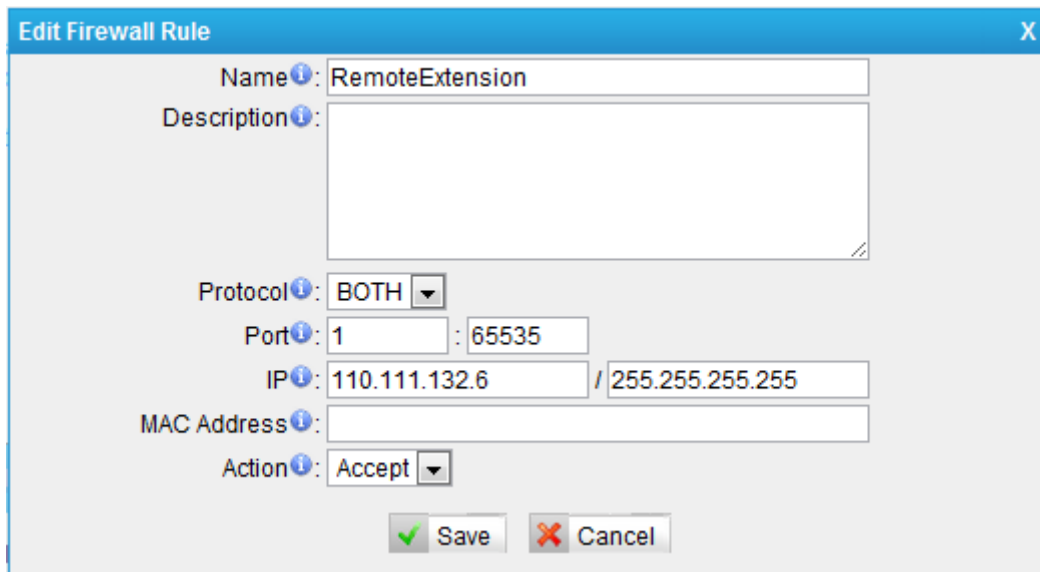


Figure G-14 Add Remote Extension IP in Whitelist

#### Step4. Configure auto blacklist rules

Auto blacklist rules: the Server would add the IP address to the blacklist automatically if the number of the packets it sends exceeds the rule you configured.

1) Add two auto blacklist rules for port: 5060.

**Rule No.1:**

**Port:** 5060

**Protocol:** UDP

**IP Packets:** 120

**Time Interval:** 60 seconds

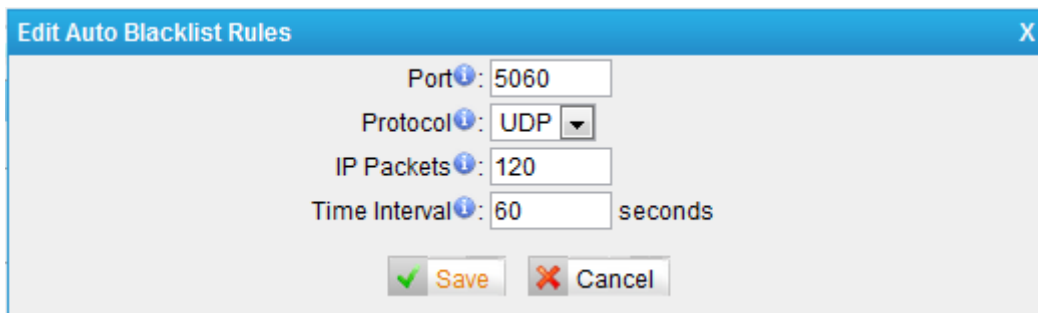


Figure G-15 Add Auto Blacklist Rule for SIP Packets

**Rule No.2:**

**Port:** 5060

**Protocol:** UDP

**IP Packets:** 40

**Time Interval:** 2 seconds

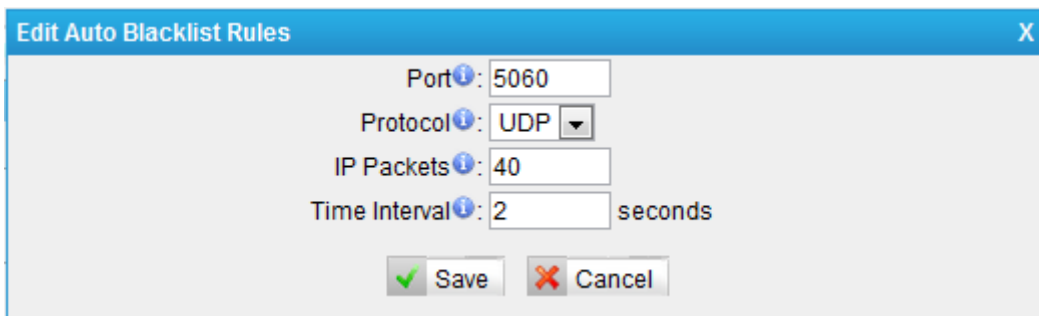


Figure G-16 Add Auto Blacklist Rule for SIP Packets

2) Add an auto blacklist rule for Port:8022

**Rule No.3**

**Port:** 8022

**Protocol:** TCP

**IP Packets:** 5

**Time Interval:** 60 seconds

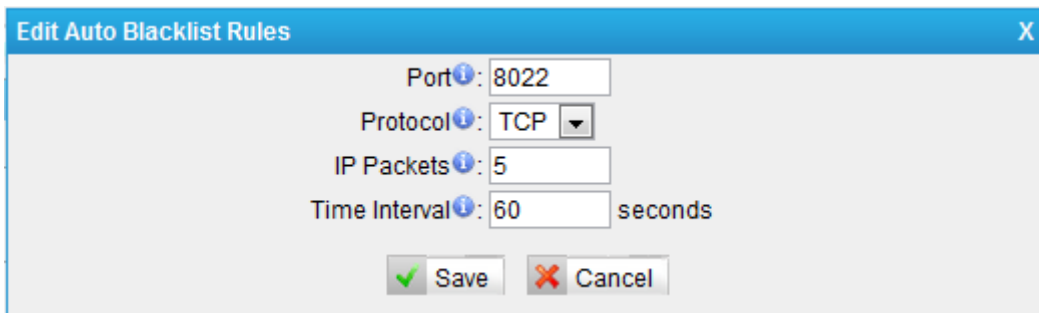


Figure G-17 Add Auto Blacklist Rule for SSH Connection

**Step 5.** Add a Firewall Rule for VoIP trunk registration

**Note:** If there is no VoIP trunk, this rule is not required. And if the RTP IP address of VoIP trunk and Registration IP address of the VoIP trunk are different, we need to create a rule to accept the RTP IP address too.

Add a rule to accept the IP address of the VoIP trunk to access MyPBX server. For example: If the IP address of the VoIP trunk is 110.5.14.6, Protocol is UDP and Port is 5060, the configuration could be as below:

**Name:** VoIPTrunk

**Protocol:** UDP

**Port:** 5060: 5060

**IP:** 110.5.14.6/255.255.255.255

**Action:** Accept

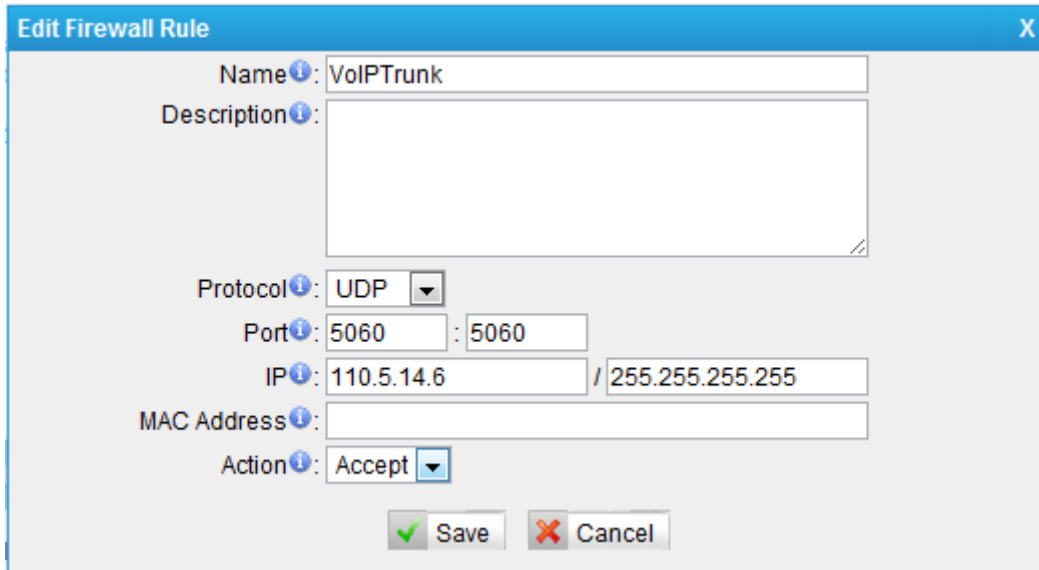


Figure G-18 Add VOIP Trunk IP in Whitelist

**Step6.** Add a firewall rule to accept the remote access of HTTP port. For example, if the remote access IP is 110.5.14.6, and the port is 80, the configuration could be as below.

**Name:** RemoteHTTP

**Protocol:** TCP

**Port:** 80:80

**IP:** 110.5.14.6/255.255.255.255

**Action:** Accept

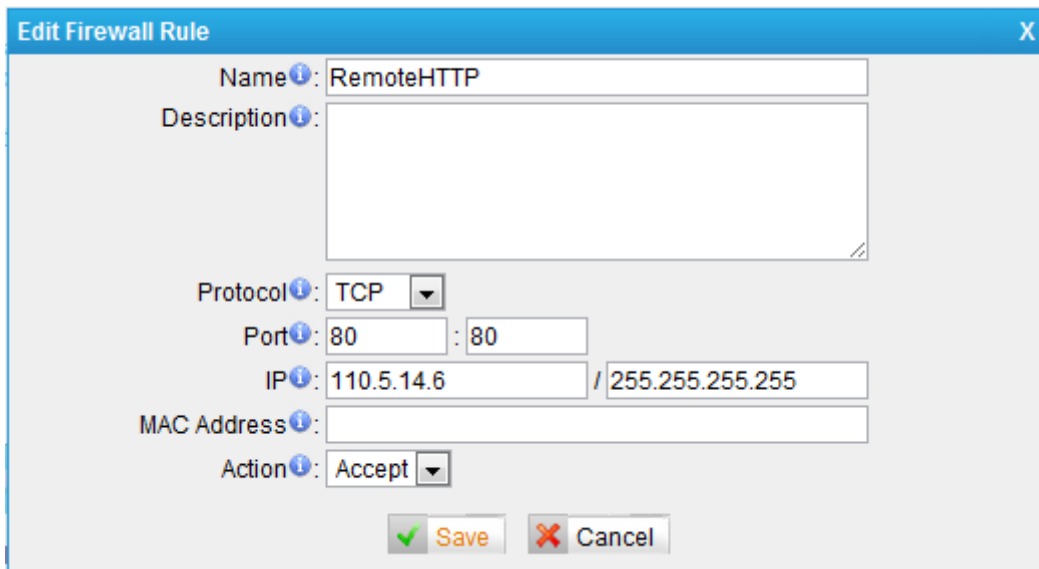


Figure G-19 Add Remote Access IP in Whitelist

**Step 7.** Add a firewall rule to accept remote access of SSH port. For example: if the remote access IP is 110.5.14.6 and the port is 8022.



**Note:** If the remote access of SSH port is not needed, this rule is not required.

**Name:** RemoteSSH

**Protocol:** TCP

**Port:** 8022:8022

**IP:** 110.5.14.6/255.255.255.255

**Action:** Accept

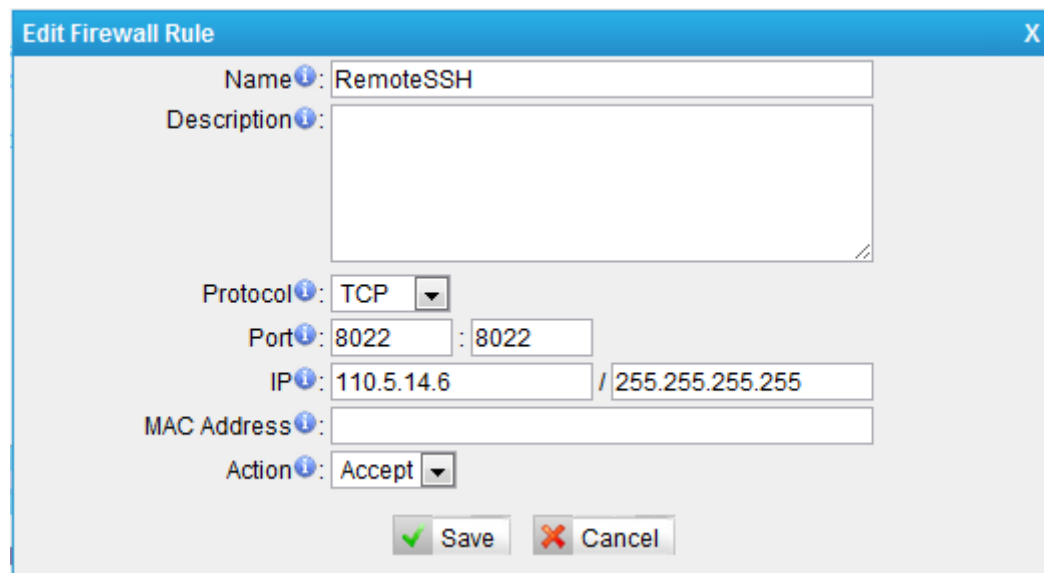


Figure G-20 Add Remote SSH Access IP in Whitelist

**Step 8.** Add other firewall rules by yourself. For example, if you are using features related to email, you should add the firewall rules for the SMTP server and POP3 server.

**Step 9.** Enable Drop all (If this feature is enabled, all the packets and connection that do not match the rules would be dropped.)

**Note:** Before enable this feature, please add a rule to accept the local network access, or the server might not be accessed.

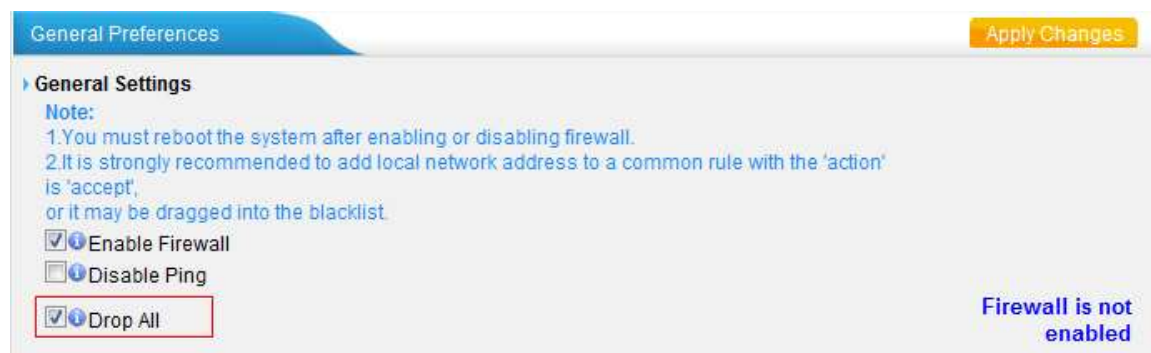
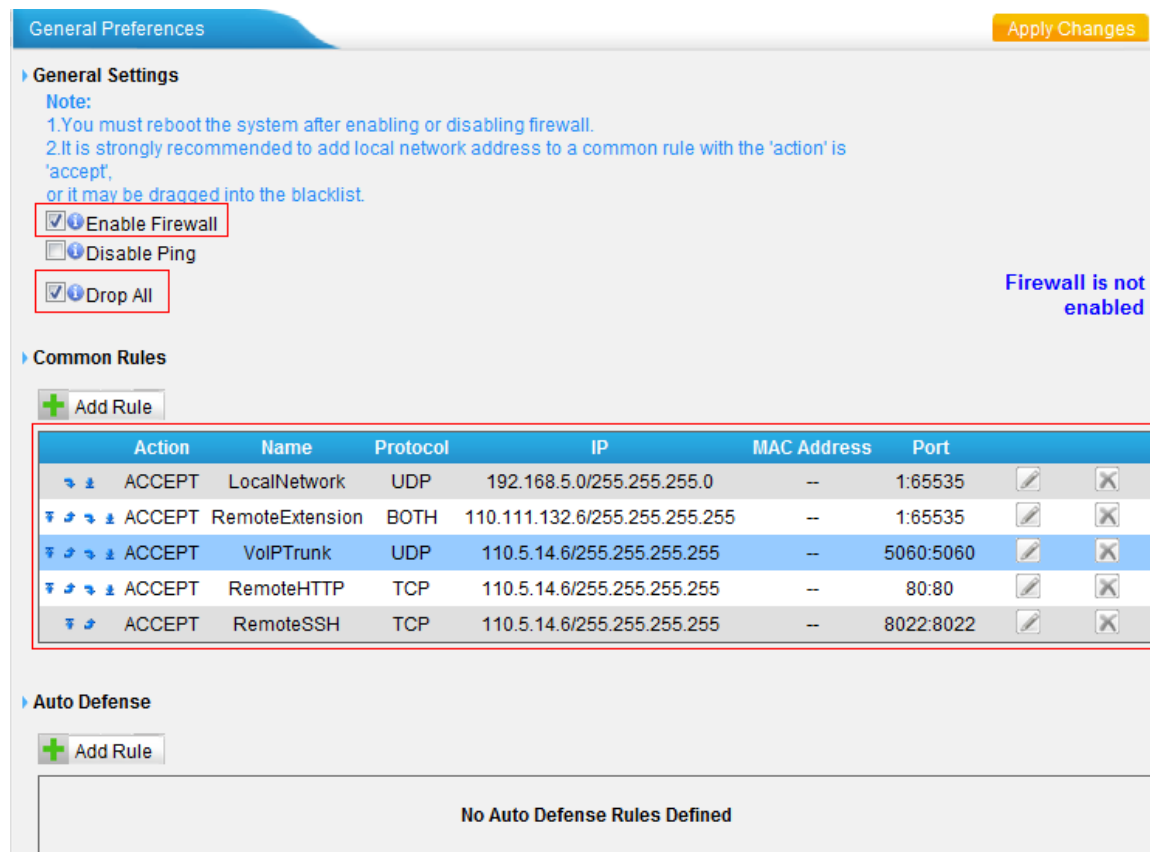


Figure G-21 Enable Drop All

**Step 10.** The Configuration of firewall settings is completed. See as below figure.



**General Preferences** Apply Changes

**General Settings**

Note:  
 1.You must reboot the system after enabling or disabling firewall.  
 2.It is strongly recommended to add local network address to a common rule with the 'action' is 'accept', or it may be dragged into the blacklist.

☒ Enable Firewall  
☐ Disable Ping  
☒ Drop All

**Common Rules**

+ Add Rule

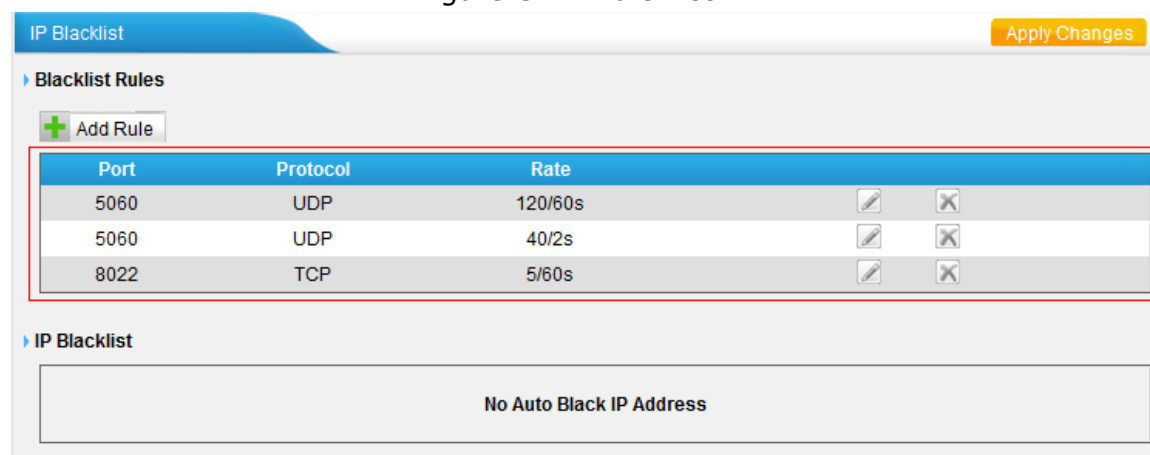
Action	Name	Protocol	IP	MAC Address	Port		
ACCEPT	LocalNetwork	UDP	192.168.5.0/255.255.255.0	--	1:65535		
ACCEPT	RemoteExtension	BOTH	110.111.132.6/255.255.255.255	--	1:65535		
ACCEPT	VoIPTrunk	UDP	110.5.14.6/255.255.255.255	--	5060:5060		
ACCEPT	RemoteHTTP	TCP	110.5.14.6/255.255.255.255	--	80:80		
ACCEPT	RemoteSSH	TCP	110.5.14.6/255.255.255.255	--	8022:8022		

**Auto Defense**

+ Add Rule

No Auto Defense Rules Defined

Figure G-22 Rule List



**IP Blacklist** Apply Changes

**Blacklist Rules**

+ Add Rule

Port	Protocol	Rate		
5060	UDP	120/60s		
5060	UDP	40/2s		
8022	TCP	5/60s		

**IP Blacklist**

No Auto Black IP Address

Figure G-23 IP Blacklist Rule List

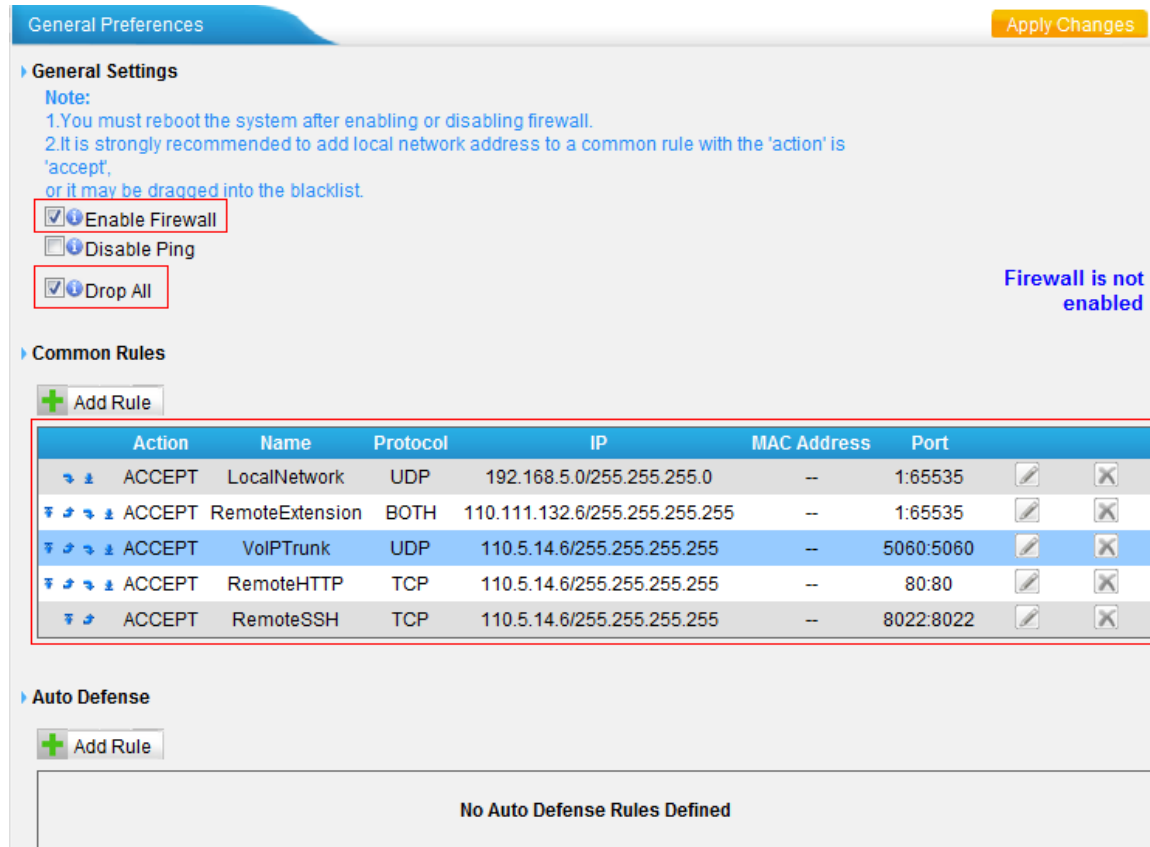
**Step 11.** Define the authorized data transmitting rate and drop all other unauthorized hosts.

**Note:** If we setup IP blacklist rules here, those unauthorized hosts will still be able to send SIP packets into MyPBX if the packets transmitting rate is under the rate defined. But as soon as the transmitting beyond the define rate, the IP will be added into blacklist automatically.

This is useful for remote extension that is using dynamic IP address.

Remove all the IP blacklist rules if you don't allowed any other unauthorized hosts access MyPBX. See the figure below.

**Note:** If 'drop all' is not enabled, please don't remove the IP blacklist rules in case the system security hole.



**General Preferences** Apply Changes

**General Settings**

Note:  
 1.You must reboot the system after enabling or disabling firewall.  
 2.It is strongly recommended to add local network address to a common rule with the 'action' is 'accept', or it may be dragged into the blacklist.

☒ Enable Firewall  
☐ Disable Ping  
☒ Drop All

**Common Rules**

+ Add Rule

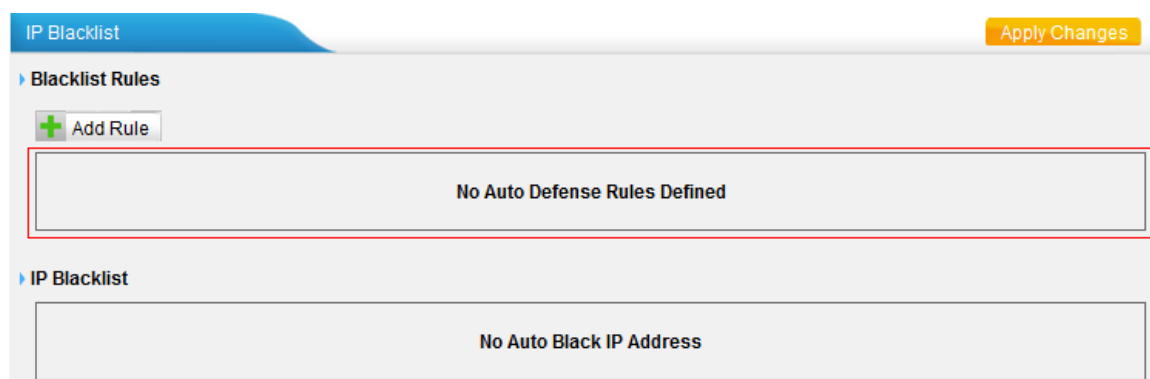
Action	Name	Protocol	IP	MAC Address	Port		
ACCEPT	LocalNetwork	UDP	192.168.5.0/255.255.255.0	--	1:65535		
ACCEPT	RemoteExtension	BOTH	110.111.132.6/255.255.255.255	--	1:65535		
ACCEPT	VoIPTrunk	UDP	110.5.14.6/255.255.255.255	--	5060:5060		
ACCEPT	RemoteHTTP	TCP	110.5.14.6/255.255.255.255	--	80:80		
ACCEPT	RemoteSSH	TCP	110.5.14.6/255.255.255.255	--	8022:8022		

**Auto Defense**

+ Add Rule

No Auto Defense Rules Defined

Figure  
G-24



**IP Blacklist** Apply Changes

**Blacklist Rules**

+ Add Rule

No Auto Defense Rules Defined

**IP Blacklist**

No Auto Black IP Address

Figure G-25

## 5. Alert Settings

After enabling alert settings', if the device is attacked, the system will alert users

via call or e-mail. The attack modes include IP attack and Web Login.

## 5.1 IPATTACK

When the system is attacked by some IP addresses, the firewall will add the IP to auto IP Blacklist and notify the user if it match the protection rule.

Example: Configure to notify extension 500, outbound number 5503301 and E-mail alert@yeastar.com.

configuration could be as below.

Phone Notification Settings:

**Phone Notification:** Yes

**Number:** 500;5503301

**Attempts:** 1

**Interval:** 60s

**Prompt:** default

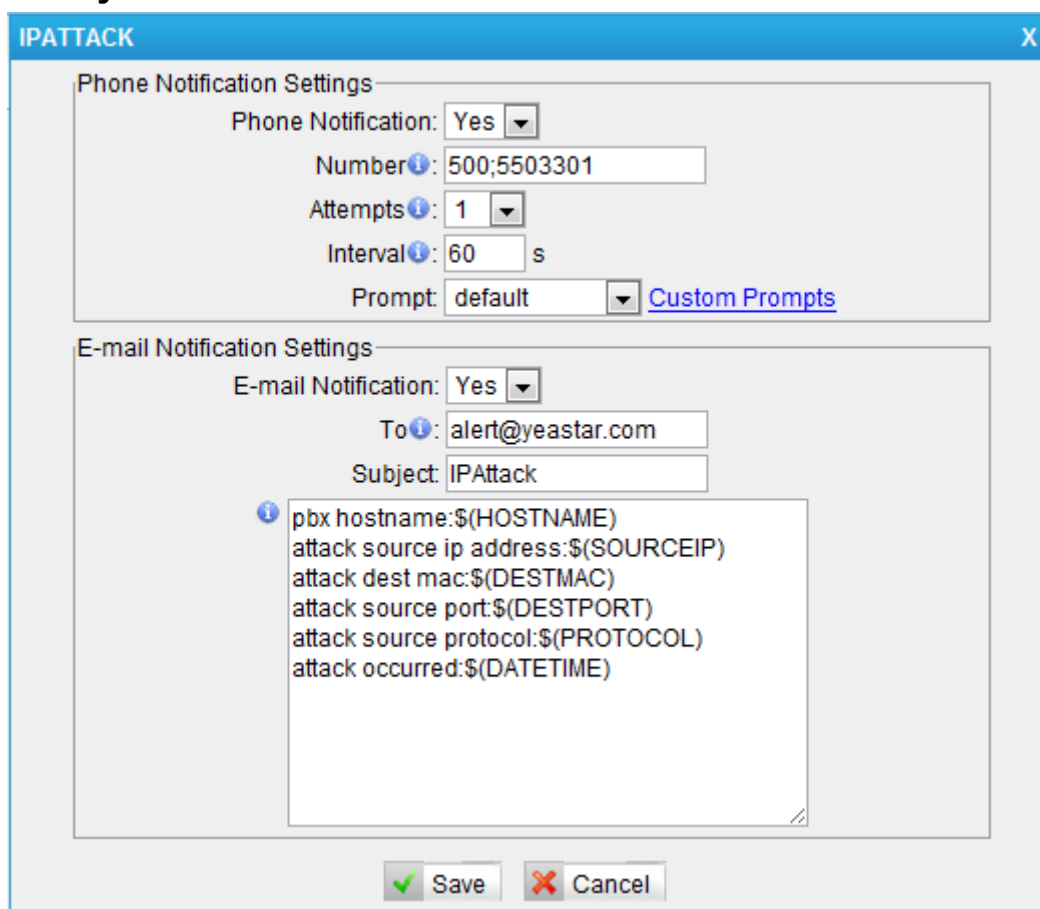
**Note:** If there's an outbound number to notify, the number should fit the dial pattern of the outbound route.

E-mail Notification Settings:

**E-mail Notification:** Yes

**To:** alert@yeastar.com

**Subject:** IPAttack



The screenshot shows a window titled "IPATTACK" with two main sections: "Phone Notification Settings" and "E-mail Notification Settings".

**Phone Notification Settings:**

- Phone Notification: Yes (dropdown)
- Number: 500;5503301 (text field)
- Attempts: 1 (dropdown)
- Interval: 60 s (text field)
- Prompt: default (dropdown) with a link to "Custom Prompts"

**E-mail Notification Settings:**

- E-mail Notification: Yes (dropdown)
- To: alert@yeastar.com (text field)
- Subject: IPAttack (text field)
- Information box (expanded):
  - pbx hostname:\$(HOSTNAME)
  - attack source ip address:\$(SOURCEIP)
  - attack dest mac:\$(DESTMIC)
  - attack source port:\$(DESTPORT)
  - attack source protocol:\$(PROTOCOL)
  - attack occurred:\$(DATETIME)

At the bottom are "Save" and "Cancel" buttons.

Figure G-26 IP Attack Notification Settings

## 5.2 WEBLOGIN

Enter the password incorrectly five times when logging in MyPBX Web interface will be deemed as attack, the system will limit the IP login within 10 minutes and notify the user.

Example: Configure to notify extension 500, outbound number 5503301 and E-mail alert@yeastar.com.

configuration could be as below.

Phone Notification Settings:

**Phone Notification:** Yes

**Number:** 500;5503301

**Attempts:** 1

**Interval:** 60s

**Prompt:** default

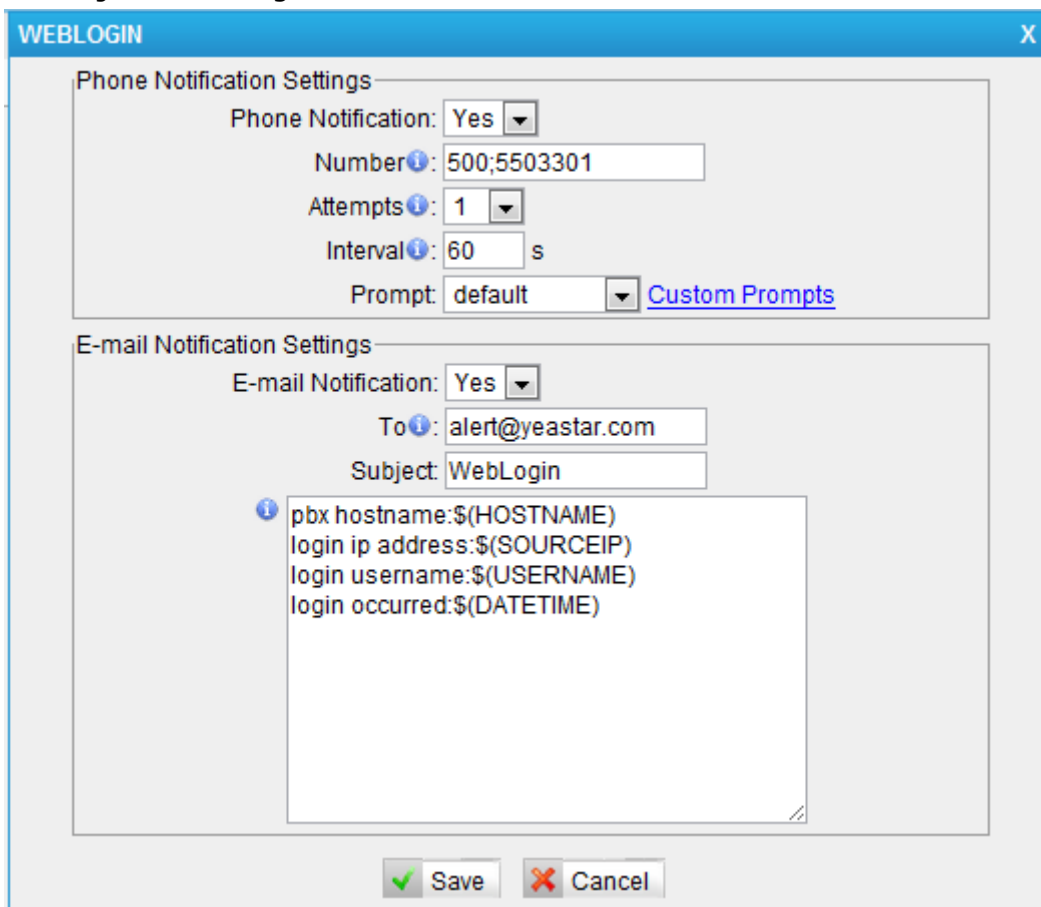
**Note:** If there is an outbound number to notify, the number should fit the dial pattern of the outbound route.

E-mail Notification Settings:

**E-mail Notification:** Yes

**To:** alert@yeastar.com

**Subject:** WebLogin



The screenshot shows a web browser window titled "WEBLOGIN" with a close button (X) in the top right corner. The window contains two main sections: "Phone Notification Settings" and "E-mail Notification Settings".

**Phone Notification Settings:**

- Phone Notification: Yes (dropdown menu)
- Number: 500;5503301 (text input)
- Attempts: 1 (dropdown menu)
- Interval: 60 s (text input)
- Prompt: default (dropdown menu) with a link "Custom Prompts" next to it.

**E-mail Notification Settings:**

- E-mail Notification: Yes (dropdown menu)
- To: alert@yeastar.com (text input)
- Subject: WebLogin (text input)
- A text area containing the following template text:
 

```
pbx hostname:${(HOSTNAME)}
login ip address:${(SOURCEIP)}
login username:${(USERNAME)}
login occurred:${(DATETIME)}
```

At the bottom of the window, there are two buttons: "Save" (with a green checkmark icon) and "Cancel" (with a red X icon).

Figure G-27 WEBLOGIN Attack Notification Settings

# APPENDIX H How to Enable Call Recording on MyPBXU300

MyPBXU300 allows you to record the calls automatically. Here is the guide for you to use it.

## Notes:

1. Before starting auto recording function, please make sure you have inserted an USB device with the correct format.
2. If you need auto recording function for MyPBXU300, please activate the call recording function by entering a license key in WEB GUI.

## 1. Activate Call Recording

**Note:** For usage of call recording on U300, please purchase activation code from your equipment provider.

Call Recording enables all your inbound and outbound phone calls to be recorded. The system would detect any call conversation and record them automatically. The call recordings would be saved in the folder you specify and stored in the USB.

And if users want to use the call recording feature, they need to active this function firstly.



Click  to enter the "Addons" page.



Figure H-1 Enable Call Recording Addon

After activation, the page will show as below:

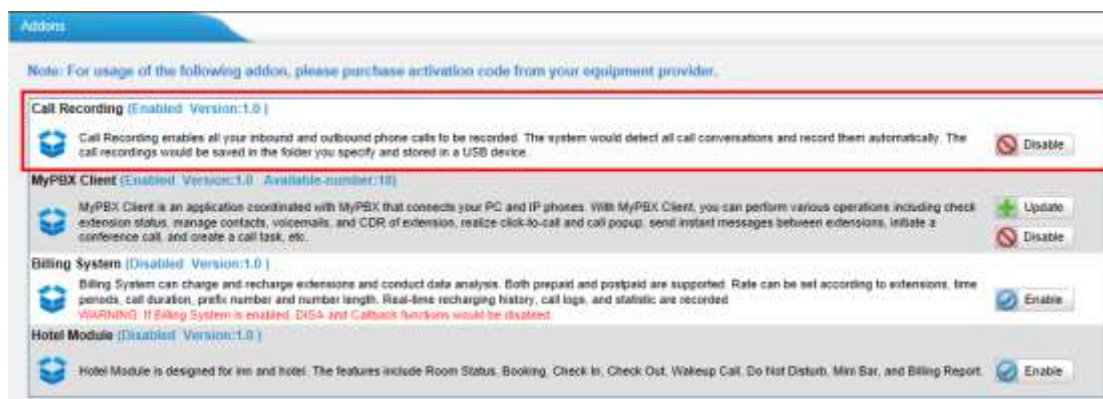


Figure H-2 Enable Call Recording Successfully

## 2. Login

From your web browser, input the IP address of the MyPBX server.  
Login the system with the monitor account:

User Name: monitor

Password: password



Figure H-3 Log In MyPBX Using monitor Account

## 3. USB Devices

### 1) USB Device information

In this page, we can check the USB Device information, such as USB Model, Type, Disk Size, Storage Info and so on.

## 2) USB Devices Management

### •Format USB

Formatting will erase all data on the USB, and initialize the USB Device.

### •Clear USB

Clearing will erase all data on the USB.

### •Remove USB Safely

When ticked, the USB Device will be removed safely.

**Warning:** Please do not remove USB device when recording a call, or else USB device will be damaged.

To remove USB device, click the button "Remove USB Safely" and follow the instructions.

### Notes:

1. When 95% of the USB device's storage space is used up, the system will automatically delete the old recordings in a chronological manner until the used space is smaller than 85%.
2. Every second of a call recording uses about 8 KB space. The following shows the storage capacity of USB device and its relevant recording hours.

Storage Capacity	2GB	4GB	8GB	16GB	32GB	64GB	128GB	256GB
Call Recording (Hours)	64	128	256	512	1024	2048	4096	8192

Figure H-4

## 4. Recording Settings

### 1) General Preferences

**Note:** Before enabling call recording, please make sure that the USB have been inserted and the format is correct.

#### •Enable Call Recording

Enable this function here. You can tick:

- a. Record Inbound calls
- b. Record Outbound calls
- c. Record Internal calls
- d. Record Callback calls.

### 2) Apply To

#### •Trunks Recording



When ticked, all calls through the selected trunks will be recorded.

### •Extensions Recording

When ticked, all calls made by the selected extensions will be recorded.

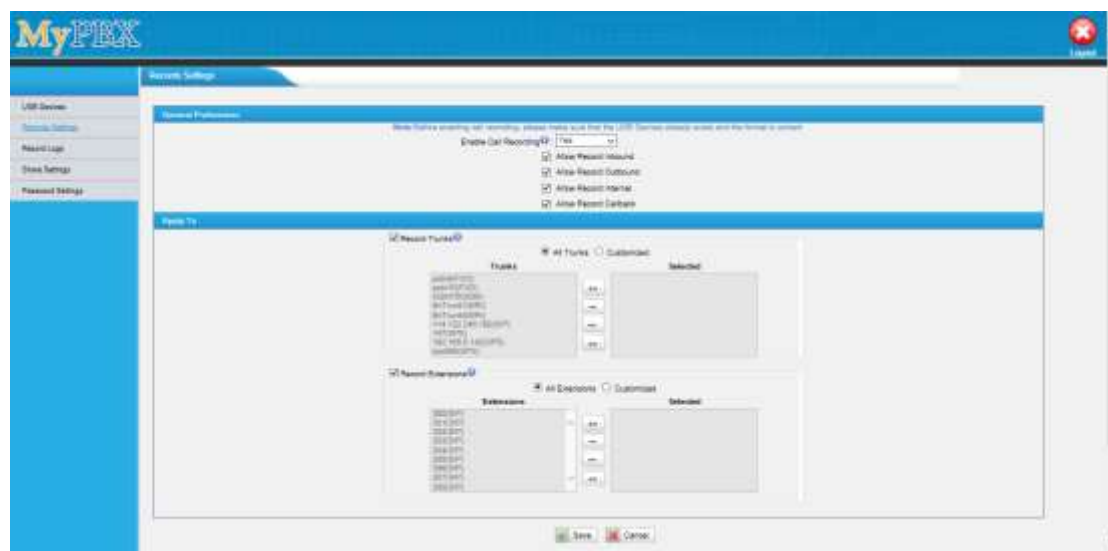


Figure H-5 Recording Settings

## 5. Record Logs

### •Start Search

You are able to search the record by Start Date to End Date, Extension, Trunk, Duration and Communication Type.

### •The Record List

You can choose the record to download or delete.

You can check the ID, Time, Callee, Caller, Trunk, Duration (second), and Communication Type of each record

You can play, download or delete the recording files.

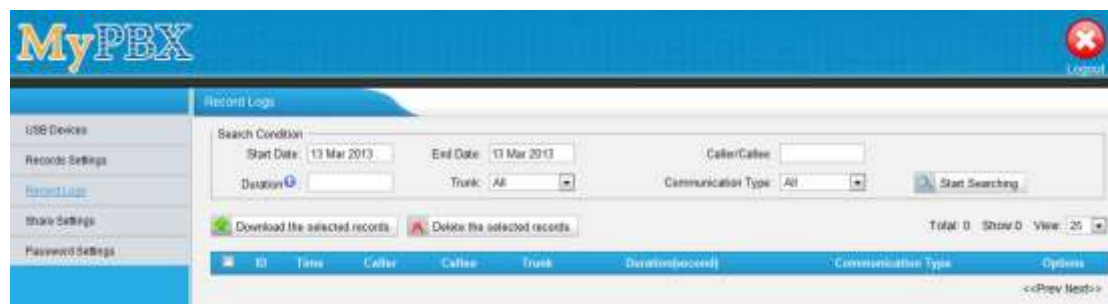


Figure H-6 Recording Logs

## 6. Share Settings

Use these settings to share the recording file on the network.

### •User Name

The user name to check the file

### •Password

The password to the user name

### •Share Name

The name of the folder showed on the network.



Figure H-7 Recording Sharing

For WIN 7/Vista user, please modify the registry before checking the shared file:

Step1: Start->Run->regedit

Step2: Modify the value of

"HKEY\_LOCAL\_MACHINE\SYSTEM\CurrentControlSet\Control\Lsa\  
LmCompatibilityLevel "  
to "1".

If "LmCompatibilityLevel" doesn't exist, create a Dword value

"LmCompatibilityLevel " = 1

You can check the recording file on the network by Explorer/Resource Manager

Run: \\ IP of MyPBX

In the picture below, the IP is 192.168.5.101

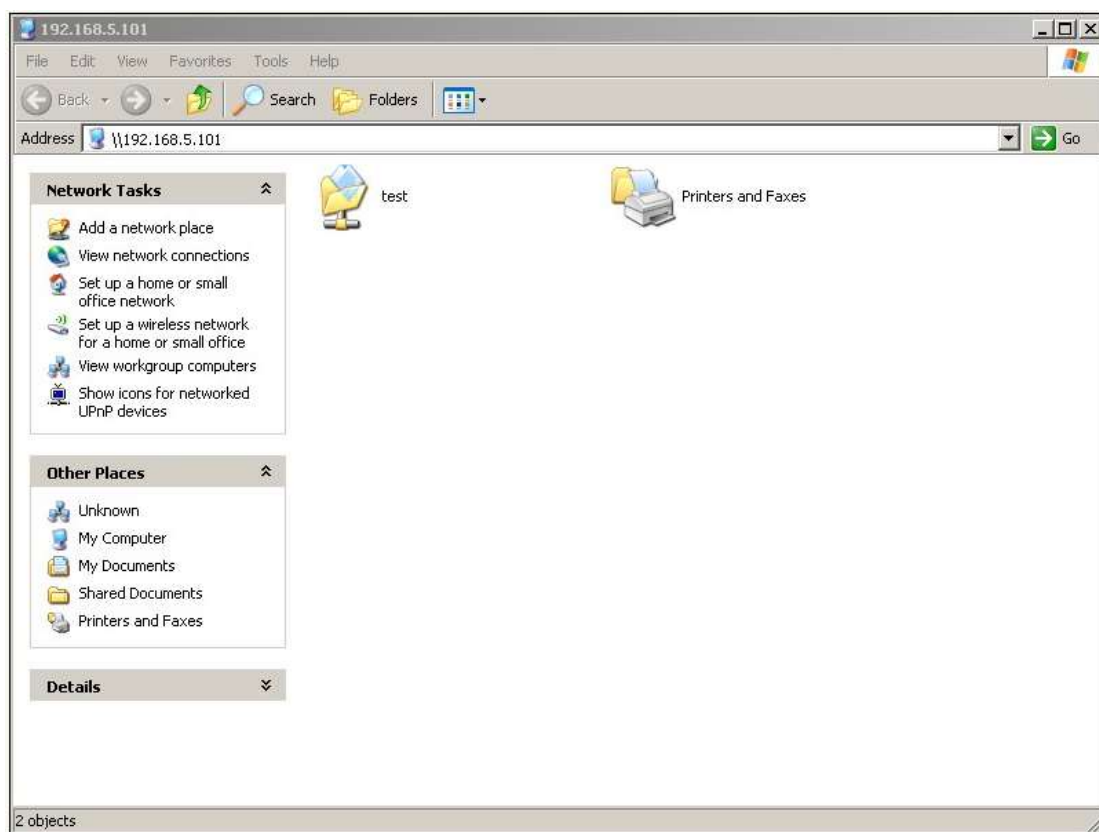


Figure H-8 Access the Sharing Folder

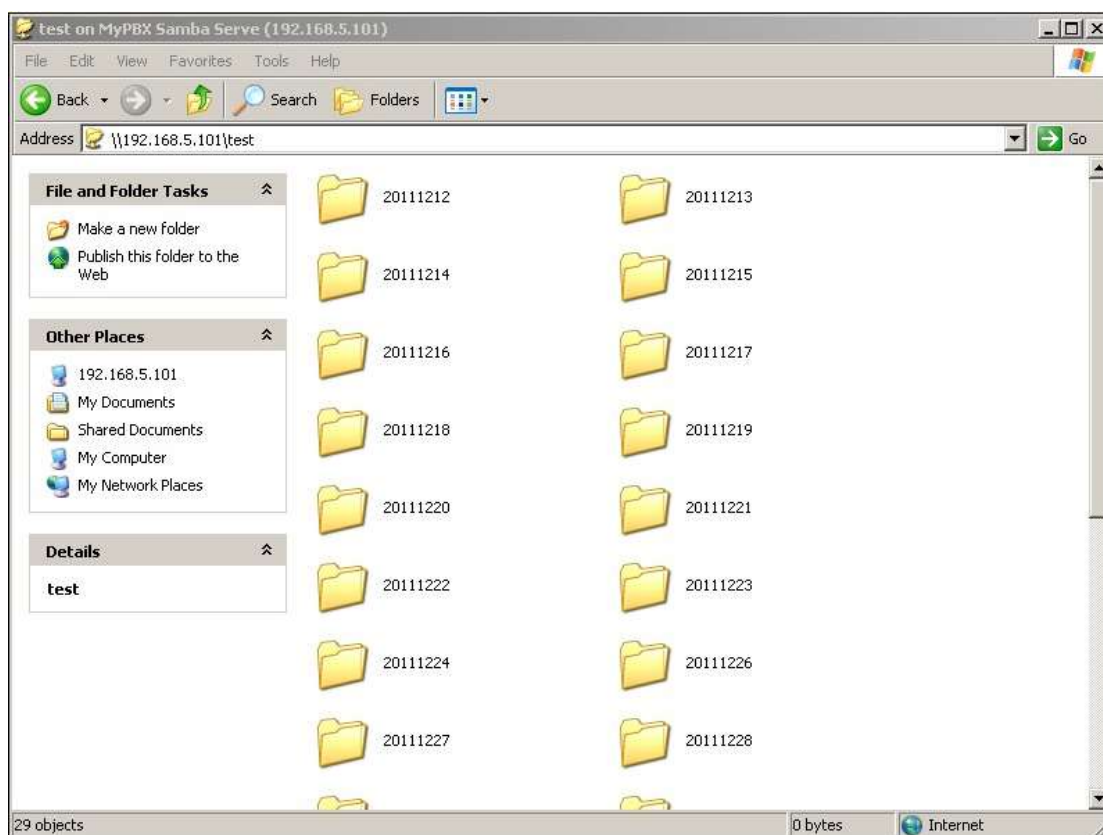


Figure H-9 Access the Sharing Folder

If you change the password of the share folder, the network users may not be able to check the shared files then.

Please ask the network user to try:

step1: Start-> Run ->cmd

step2: net use \* /del (there is a space behind \*)

step3: Visit the device again with new password

## 7. Password Settings

You can change the password of the account "monitor" here.



Figure H-10 Modify monitor Password

<Finish>

# MyPBX U300 Extension User's Guide

Version:V1.1

**Yeastar Technology Co., Ltd.**

Date: 2013.8.9

# Contents

<b>INTRODUCTION .....</b>	<b>3</b>
<b>1. MYPBX USER WEB INTERFACE .....</b>	<b>3</b>
1.1 LOGIN .....	3
1.2 CALL SETTINGS .....	4
1.2.1 <i>Follow Me Settings</i> .....	4
1.2.2 <i>Call Waiting</i> .....	5
1.2.3 <i>Do not Disturb (DND)</i> .....	5
1.2.4 <i>Ring Out</i> .....	5
1.2.5 <i>Mobility Extension Number</i> .....	5
1.3 VOICEMAIL .....	7
1.3.1 <i>Voicemail Settings</i> .....	7
1.3.2 <i>Voicemail Management</i> .....	7
1.4 CALL RECORDINGS .....	9
<b>2. CONCLUSION.....</b>	<b>9</b>

## INTRODUCTION

Thanks for using MyPBX U300 User Web Interface. This Guide would explain how to configure the “Follow me” settings (in different cases such as, “No Answer”, or “When Busy”; whether transfer the incoming call to voicemail or other phone number); how to configure Voicemail settings (such as enable/disable voicemail service, set a voicemail password and enable/disable “send the voicemail to an email address as an attachment” service); how to manage the extension's voicemail and call recordings.

## 1. MYPBX USER WEB INTERFACE

### 1.1 Login

The default user name and password to login the User Web Interface:

User Name: Extension number

Password: Voicemail Access PIN#

Note: If it's unable to log in with default user name and password, please ask the administrator to check if you have the authority to log in, or have the right user name and password.

**Hybrid IP PBX for Your Businesses**

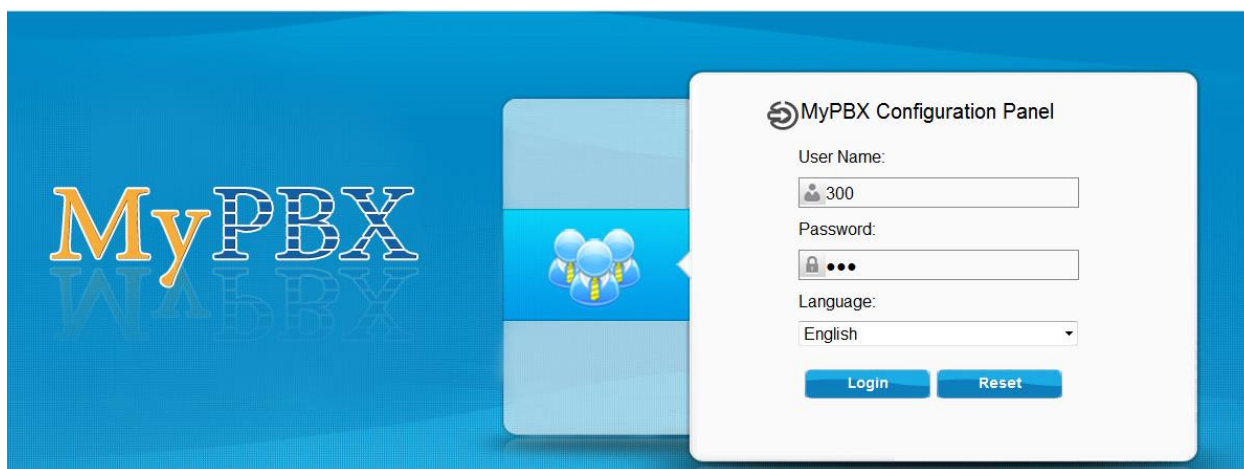


Figure 1-1 Login Page

## 1.2 Call Settings

In the Call Settings page, you could configure features like “Follow me”, “Call waiting”, “DND”, and “Ring Out”.

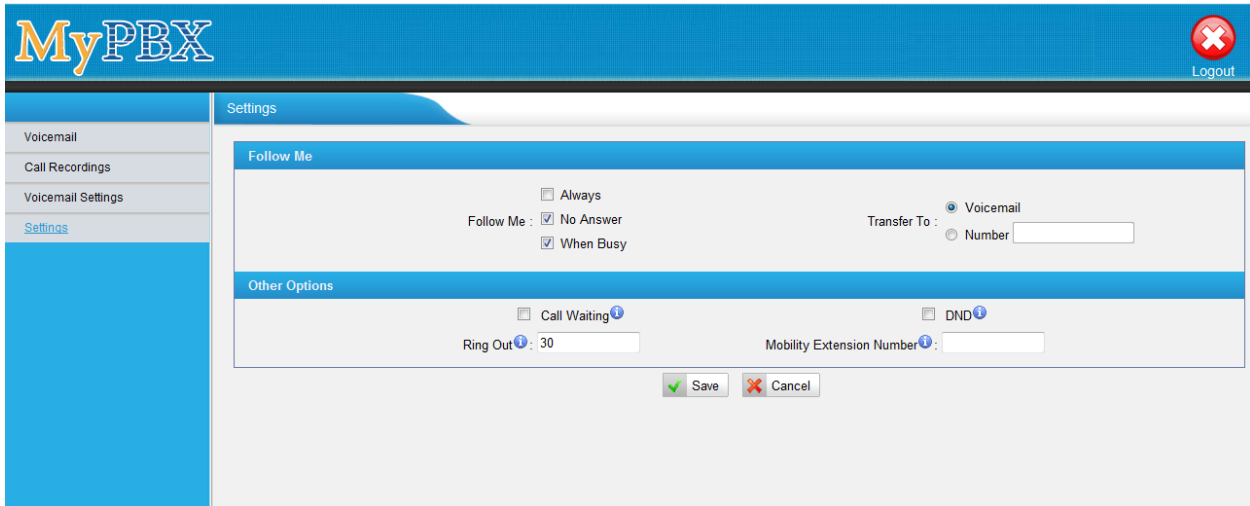


Figure 1-2 Extension Settings

### 1.2.1 Follow Me Settings

#### 1. Always

**Always Follow Me** is a service of MyPBX that allows the automatic unconditional transfer of incoming calls to varying other destinations.

Enable **Always** checkbox selection activates the unconditional call forwarding service on the current extension. The **Transfer To** displays the destinations where the incoming call from a particular caller should be unconditionally forwarded to.

- Transfer To** choose voicemail, the incoming call will be unconditionally forwarded to voicemail.
- Transfer To** choose Number, the incoming call will be unconditionally forwarded to other extension, PSTN or mobile phone number.

**Note:** When **Always Follow Me** configuration is saved, the Ring Group and Call Queue services will be disabled.

#### 2. No Answer

**No Answer Follow Me** is a service of MyPBX that allows the automatic transfer of incoming calls to varying other destinations when the called extension is unable to pick up the call. No answer timeout is configured as the **Ring Out**.

Enable **No Answer** checkbox selection activates no answer call forwarding service on the current extension. The **Transfer To** displayed the destinations where the incoming call from a particular caller should be



forwarded to if the called extension is unable to answer the call.

- a. **Transfer To** choose voicemail, the incoming call will be forwarded to voicemail when there is no answer.
- b. **Transfer To** choose Number, the incoming call will be forwarded to extension, PSTN or mobile phone number when there is no answer.

**Note:** When **No Answer Follow Me** configuration is saved, the Ring Group and Call Queue services will be disabled.

### 3. When Busy

**When Busy Follow Me** is a service of MyPBX that allows the automatic transfer of incoming calls to varying other destinations when the called extension is busy with another call.

Enable **Busy** checkbox selection activates the busy call forwarding service on the current extension.

The **Transfer To** displayed the destinations where the incoming call from a particular caller should be forwarded to when the called extension is busy.

- a. **Transfer To** choose voicemail, the incoming call will be forwarded to voicemail when the callee is busy.
- b. **Transfer To** choose Number, the incoming call will be forwarded to other extension, PSTN or mobile phone number when the callee is busy.

**Note:** When saving the **When Busy Follow Me** configuration, the Ring Group and Call Queue services will be disabled.

### 1.2.2 Call Waiting

**Call Waiting** is used to receive an incoming call when already on a call. A ringing tone will be heard by the caller and a special beeping will be heard on the phone by the MyPBX user when the call arrives. To switch between the current call and the new incoming call use the appropriate calling code.

Enable **Call Waiting** checkbox selection activates this service and makes it available for the phone's handset.

Note: If the Call Waiting service is enabled, the Busy Call Forwarding service would be disabled automatically.

### 1.2.3 Do not Disturb (DND)

Do Not Disturb service is used to turn off the phone bell. If it's enabled, incoming callers will be routed to the voicemail box (if enabled); otherwise they will be disconnected automatically.

### 1.2.4 Ring Out

Ring Out is used to define the ringing time of the extension. If the ringing time exceeds the configured time, the extension would stop ringing.

### 1.2.5 Mobility Extension Number

Mobility Extension Number is used to enable the cell phone to work as a mobile extension. If a cell phone number is set as the mobility extension number, once it calls MyPBX, the cell phone would have all authority

of extension. In this way, the user is able to check the voicemail, call the other extension, PSTN or mobile numbers via cell phone.

## 1.3 Voicemail

Voicemail is used for the caller to leave a message to the called extension when it is unavailable or busy. The voicemail would be stored in the extension's voicemail box. When the caller accesses the voicemail service, he will hear a voice prompt that remind him to leave the message (This voice prompt could only be configured by the administrator). The max duration of each voicemail is 15 minutes (The default value is 5 minutes).

### 1.3.1 Voicemail Settings

In the Voicemail Settings page, the extension user is able to set the password for voicemail access (which is the login password of User Web Interface as well); Enable or disable to the voicemail service; enable or disable "send the voicemail to an email address as an attachment" service.

**Set a new voicemail password:** Enter the new password, retype it and click "save".

**Email Address:** Enter the email address where the voicemail or fax would be sent to as an attachment.

**Enable Voicemail:** Check the box to enable the voicemail service or uncheck to disable.

**Enable Send Voicemail:** Check the box to enable "send the voicemail to an email address as an attachment" service or uncheck to disable.

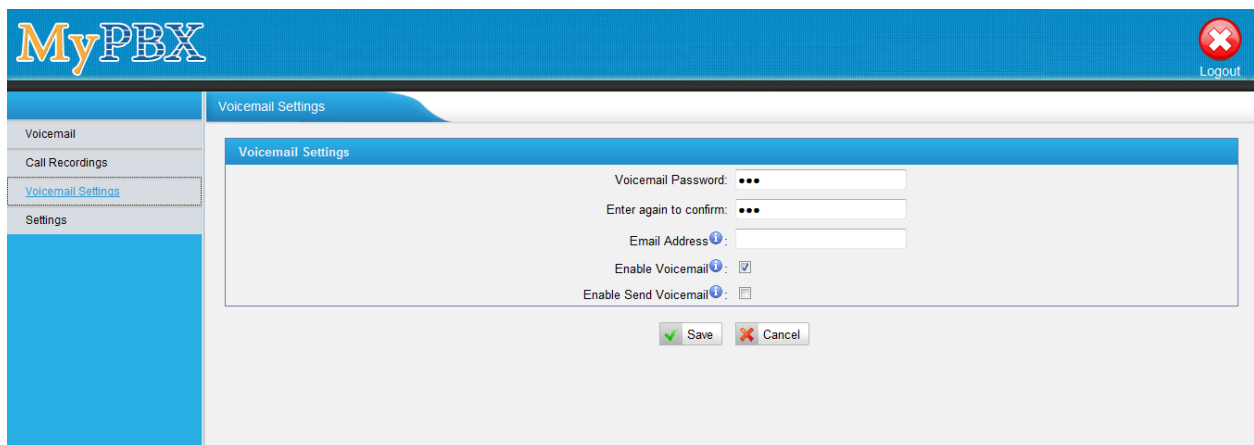


Figure 1-3 Modify Voicemail Settings

### 1.3.2 Voicemail Management

In the Voicemail Management page, the extension user could play, delete or download the voicemail, or move the voicemail from one folder to the other folder (e.g. New, Old, Family, Work, Friends).

**Play the voicemail:** Select the voicemail the play and click "Play". In the "Play" page, the user could control the speed and volume.

**Delete the voicemail:** Select the voicemail to delete and click "Delete", then the selected email would be deleted.

**Download the voicemail:** Select the voicemail to download and click "download".

**Move the voicemail to another folder:** Select the voicemail to move, select the folder to move to (e.g. New, Old, Family, Work, Friends), and then click "move to".

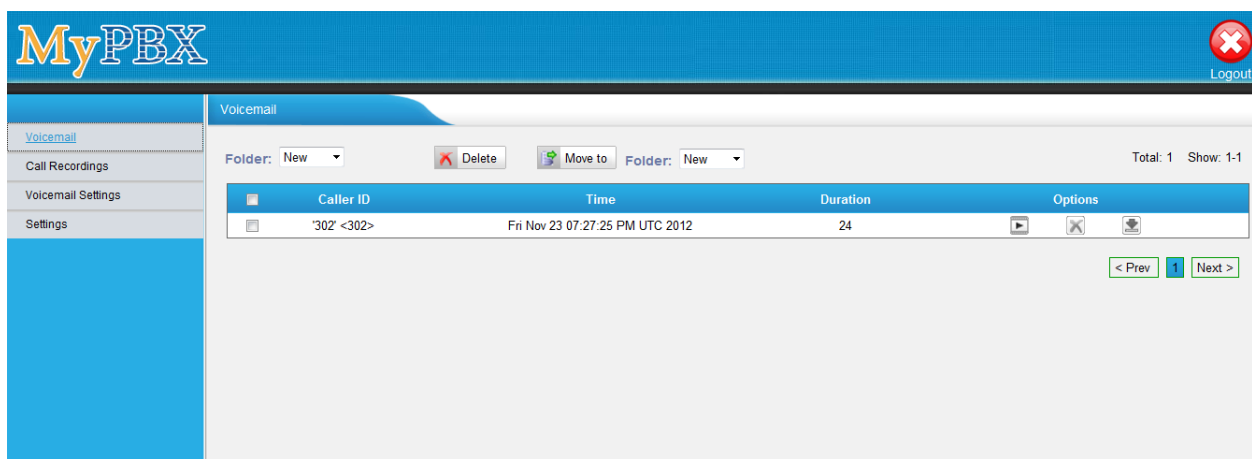


Figure 1-4 Voicemail Management

## 1.4 Call Recordings

In the Call Recording page, the user could play, delete and download the recording files.

**Play the recording:** Select the recording the play and click “Play”. In the “Play” page, the user could control the speed and volume.

**Delete the recording:** Select the recording to delete and click “Delete”, then the selected email would be deleted.

**Download the recording:** Select the recording to download and click “download”.

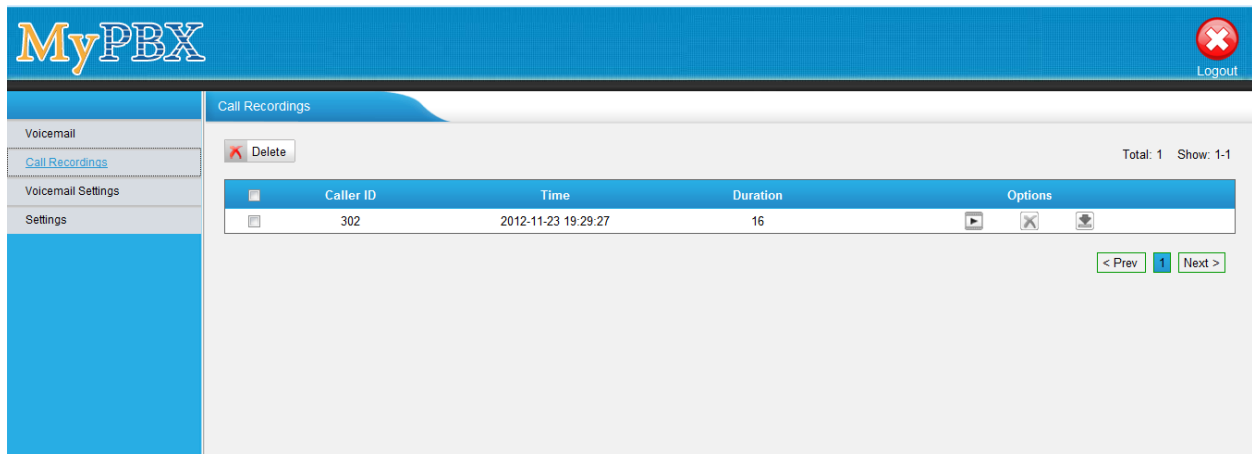


Figure 1-5 Recording Management

## 2. CONCLUSION

This User's Guide only explains the configuration of call settings and voicemail settings; management of voicemails and recordings for extension users. For more functionality and advanced settings of MyPBX U300, please refer to the relative documents as below:

***“MyPBX U300 Datasheet”***

***“MyPBX U300Quick Installation Guide”***

***“MyPBX U300 Administrator's Guide”***

[The End]